30 MARCH 1979 (FOUO 18/79) 1 OF 2

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30 March 1979

USSR

TRANSLATIONS ON USSR SCIENCE AND TECHNOLOGY
PHYSICAL SCIENCES AND TECHNOLOGY
(FOUO 18/79)
EIGHTH ALL-UNION SEMINAR ON STATISTICAL HYDROACOUSTICS

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TRANSLATIONS ON USSR SCIENCE AND TECHNOLOGY
PHYSICAL SCIENCES AND TECHNOLOGY
(FOUO 18/79)

EIGHTH ALL-UNION SEMINAR ON STATISTICAL HYDROACOUSTICS

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKA in Russian 1977 signed to press 14 Dec 77, pp 3-6, 21-23, 26-27, 29-32, 45-61, 66-76, 78-93, 98-123, 127-164, 169-170, 175, 177-184, 187-189

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CONTENTS (Continued) Page

GEOPHYSICS, ASTRONOMY AND SPACE

UDC 534.288

RELATIONSHIP OF LOW-FREQUENCY NOISE LEVELS OF THE SEA TO THE HYDROMETEOROLOGICAL SITUATION

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 3-6

[Article by Ye. P. Masterov, S. F. Cherepantsev and S. P. Shorokhova]

[Text] Despite the large number of publications devoted to experimental measurement of low-frequency noise levels of the sea, the question of noise level variation on the time of day and season has been inadequately studied. This paper, in which the causes of the indicated variations are analyzed, is devoted to this problem.

Systematic measurements of the spectral-energy and hydrometeorological characteristics at various times of the day and different times of the year were carried out to study the variability of the spectral low-frequency noise levels of the sea. Experimental materials obtained in February, June, July and September are presented in the given paper. Observations of sea noise levels were made in February only during 3 days, while those made during the remaining months cover a period of 10 days. The measurements were made by using a bottom hydrophone installed in a region with depth of approximately 200 m at a distance of 2 km from the shoreline. The hydrophone was placed in a deflector. The measured sea noise levels exceeded the natural noise levels of the pre-amplifier by less than 6 dB. The hydrometeorological characteristics in the region of conducting the experiment were recorded simultaneously with measurements of sea noise. The results of full-scale measurements were processed on the "BESM-4M" computer.

The dependence of the main hydrometeorological characteristic on the time of day for the most typical seasons of the year are presented in Figure 1. The Roman numerals on the curves denote the number of the month. As can be seen from the curves shown in Figure 1, a, the greatest variability of wind velocity over a 24-hour period occurred in June. The daily variations of sea wave height, water temperature and air-water temperature gradient are shown in Figure 1, b, c and d. It is easy to see from the dependence of water temperature on the time of day given in Figure 1, c that the heat content of the surface layer increases gradually toward fall regardless of

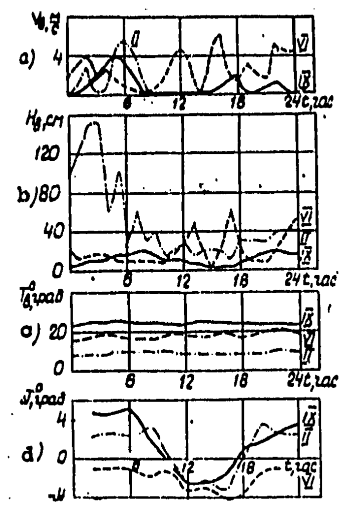

Figure 1. Dependence of Wind Velocity (a), Wave Height (b), Water Temperature (c) and Water-Air Temperature Gradient (d) on Time of Day

the time of day. The water-air temperature gradient (Figure 1, d) has an extreme value in the region of 1200-1600 and is more clearly marked for summer and fall months.

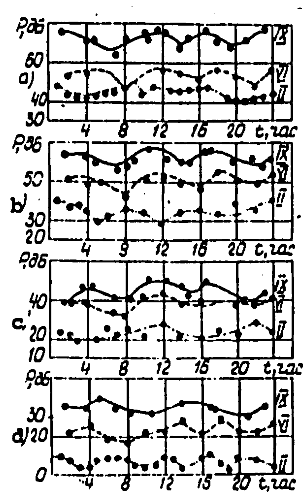

Figure 2. Dependence of Sea Noise Levels at Frequencies of 2 Hz (a), 20 Hz (b), 200 Hz (c) and 2,000 Hz (d) on Time of Day

The daily variations of the noise levels reduced to the 1-Hz band, in decibels with respect to pressure of $2\cdot10^{-5}$ n/m^2 for frequencies of 2, 20, 200 and 2,000 Hz, are presented in Figures 2, a, b, c and d, respectively. The Roman numerals on the graphs denote the month. An increase of noise levels upon transition from winter to summer and fall is typical for all the given functions. This is apparently caused by the increase of heat content of the surface layer of the sea, noted above, for the given region of measurements. Periodic fluctuations of noise levels with the greatest maximum at approximately 1200 are observed over a 24-hour period. Besides a 24-hour period, periods of fluctuations at intervals of 1200 and 0600-0800 are considered. The range of noise level fluctuations for each month reaches 8-10 dB. The autocorrelation functions of noise levels at frequencys of 2, 20, 200 and 2,000 Hz are presented in Figures 3-6. As can be seen, 24-hour periodicity is clearly marked for all frequencies during summer, especially in June. The most interesting from our viewpoint is determination of a frequency dependence of the values of the correlation relation between low-frequency noise levels of the sea and such hydrometeorological parameters as wind velocity, sea wave height, water temperature and the water-air temperature gradient. The frequency dependence of the cross-correlation coefficient between spectral noise levels of the sea and the enumerated hydrological characteristics are shown in Figure 7, a, b, c and d. The value of the correlation relation of noise levels to wind velocity and wave height according to the frequency range varies over a wider range in September (Figure 7, a) than in February. As indicated by the given analysis, variations of sea noise levels are related to a greater extent to variations of water temperature and of the water-air temperature gradient than to variations of wind velocity and sea state.

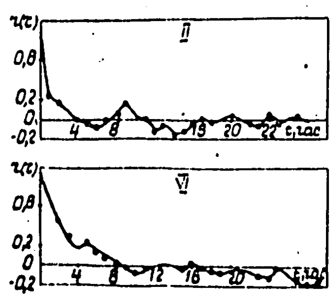

Figure 3. Autocorrelation Characteristics of Sea Noise Levels at Frequency of 2 Hz

The experimental investigations indicate the presence of a dependence of low-frequency noise level variations of the sea on different hydrological parameters, including those due to the heat content of the surface layer of the sea.

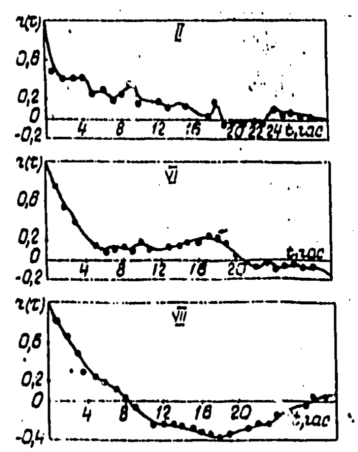

Figure 4. Autocorrelation Characteristics of Sea Noise Levels at Frequency of 20 Hz

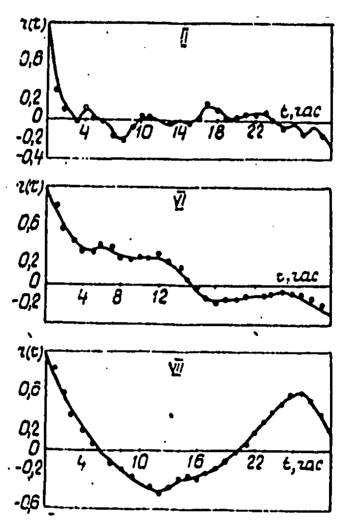

Figure 5. Autocorrelation Characteristics of Sea Noise Levels at Frequency of 200 Hz

It must be noted that the given results are preliminary. Further investigations are required to reach final conclusions.

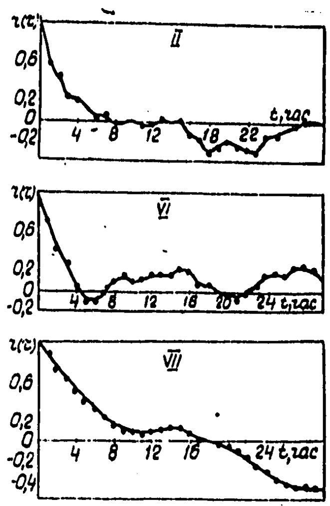

Figure 6. Autocorrelation Characteristics of Sea Noise Levels at Frequency of 2,000 Hz

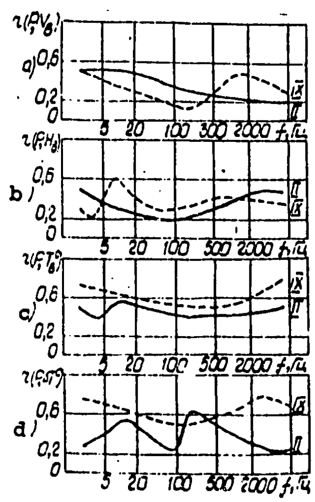

Figure 7. Frequency Dependence of Cross-Correlation Coefficient of Sea Noise Levels on Wind Velocity (a), Wave Height (b), Air Temperature (c) and Water-Air Temperature Gradient (d)

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GEOPHYSICS, ASTRONOMY AND SPACE

UDC 534.88

INVESTIGATING THE METHOD OF DETERMINING THE ANISOTROPY OF THE NOISE FIELD

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 p 18

[Article by L. A. Bespalov, A. M. Derzhavin, O. L. Sokolov and M. L. Turbovich]

[Text] The noise level is measured in several sectors of space to determine the anisotropy of the hydroacoustic field by a detector located at one point of the investigated space. Further analysis and decision-making are carried out on the basis of the Cochran logarithmic criterion

$$\mu = \ln \frac{\sum_{i=1}^{M} S_i^2}{\max S_i^2} ,$$

where

$$S_i^2 = \sum_{j=1}^{L} x_{ij}^2$$

is the estimate of dispersion in the i-th sector of observation by L independent readings in M observation sectors.

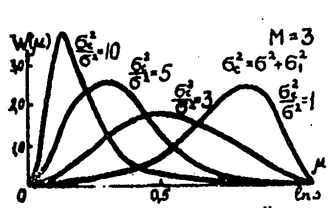

Figure 1. Differential Distribution Law

The Cochran criterion was proposed in its time to test a number of dispersions for uniformity [1]. There are approximate calculations in this case which permit one to determine the threshold points for μ, corresponding to the probability of a false alarm on the order of 5 and 1 percent. We obtained the distribution of the Cochran criterion μ in the case applied to an isotropic field (signal dispersion in the sector σ^2) of a signal source (with dispersion σ_1^2) in one of the sectors, which permits one to determine the probabilities of correct detection of the signal source on a background of isotropic noise. The working characteristics of detection are presented in Figure 2.

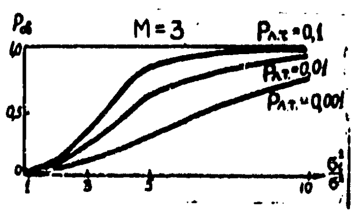

Figure 2. Working Characteristics

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GEOPHYSICS, ASTRONOMY AND SPACE

UDC 551.463.2

THE CAPACITY OF THE HYDROACOUSTIC COMMUNICATIONS CHANNEL IN THE PRESENCE OF SCATTERING ON THE SEA SURFACE

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 21-22

[Article by Ya. P. Dragan and I. N. Yarovskiy]

[Text] For communications capacity with scattering on the sea surface, when the length of the surface wave L is much greater than the acoustic wave λ, following [1-4], using the results of [5], we find:

$$C_1 = \frac{1}{4\pi}\int_{\omega_0-W}^{\omega_0+W} \log\left\{\left(\frac{P_c}{P_{ш}} - \frac{1}{8Kc_0(z-\sum_l A_l\cos(l\nu t))}\left[\operatorname{ctg}(k_0+K)c_0(z-\sum_l \cos(l\nu t)) - \operatorname{ctg}(k_0-K)c_0(z-\sum_l A_l\cos(l\nu t))\right]\right)4\sin^2 kc_0(z-\sum_l A_l\cos(l\nu t))\right\}d\omega.$$

Here A_l are the amplitudes of the harmonic components of the surface. If the inequality $L \gg \lambda$ is not fulfilled, the capacity is determined with an accuracy up to first order by the expression

$$C_2 = \frac{1}{4\pi}\int_{\omega_0-W}^{\omega_0+W} \log\left\{\left[\frac{P_c}{P_{ш}} + \frac{1}{8W}\int_{\omega_0-W}^{\omega_0+W}\frac{d\omega}{\sin^2 kc_0 z - kc_0\sin kc_0 z\sum_l A_l\cos(l\nu t-\gamma_l)}\right] \times \left[4\sin^2 kc_0 z - 4kc_0\sin kc_0 z\sum_l A_l\cos(l\nu t-\gamma_l)\right]\right\}d\omega,$$

where

$$\gamma_l = k[(a_l - a_0)x + c_l z],\quad a_l = a_0 + l\frac{\lambda}{L},\quad c_l = (1-a_l^2)^{\frac{1}{2}}.$$

Using the last formulas for some parameters of a communications channel and input signal, estimates of the effect of a plane surface wave on capacity and the dependence of capacity on time when a regular wave is propagating along the surface are calculated.

Calculation shows that reflection from the surface may lead both to an increase and a decrease of it as a function of the depth of submersion z. The relative effect of the surface is reduced and the capacity asymptotically approaches some value distinct from this value in the absence of reflection by some constant value dependent on z with an increase of the ratio P_s/P_{sh}.

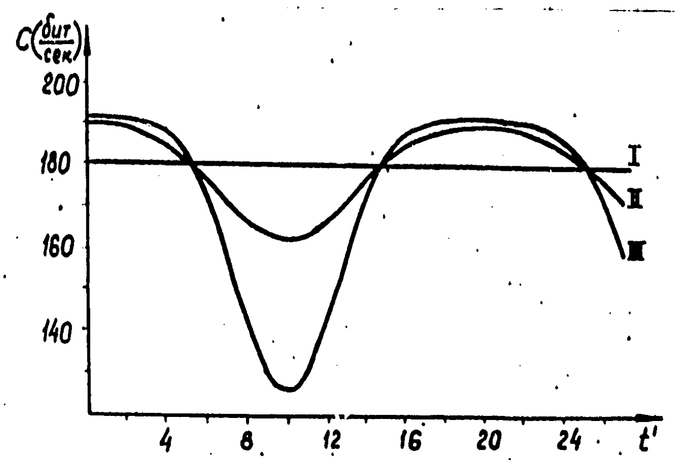

Figure 1. Dependence of Channel Capacity on Time at $kc_0z = 2.05$, $P_s/P_{sh} = 50$, $F = 25$ Hz; I -- $A_1 = 0$; II -- $kc_0A_1 = 0.31$; III -- $kc_0A_1 = 0.62$

Wave action of the water surface leads to variation of capacity with time, the nature and depth of which, as follows from Figure 1, is significantly determined by the value of surface roughness.

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GEOPHYSICS, ASTRONOMY AND SPACE

UDC 37.02.05.01.07

DETERMINING THE CAPACITY OF A HYDROACOUSTIC COMMUNICATIONS CHANNEL

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 p 23

[Article by V. S. Nesterov]

[Text] A formula is derived in the paper for calculation of the lower bound of the capacity of a hydroacoustic channel with regard to restriction on peak power, frequency dependence of vibration attenuation and sea noise spectrum and measurement of signals at the reception point. The Shannon formula was used in this case with restriction on the peak power and white noise, Knudsen curves, the LeBlanc formula [1] and the author's method [2] for finding the capacity of a communications channel with fading of the equal capacity of the channel without fading with white noise by using an adaptive predictive filter in the detector which compensates for the frequency characteristic of the channel in the fading frequency band. The final formula has the form:

$$C \geq \Delta f \log_2 \left[\frac{\frac{2}{\pi e^{2}} \cdot \frac{S \cdot f^{n}}{N_{1kHz}} \cdot \frac{m}{r^{2}} + 1}{\frac{2}{\pi e^{2}} \cdot \frac{S \cdot f^{n}}{N_{1kHz}} \cdot \left(m - \frac{1}{r^{2}} \right) + 1} \right],$$

where C is the capacity, S is the permissible emitted peak power, N_{1kHz} is the noise level in the 1-Hz band at frequency of 1 kHz with the corresponding sea state expressed in bals, f is frequency (kHz), Δf is the frequency bandwidth of the detector (Hz), m is the number of beams at the reception point and r is the distance from the source to the detector.

The capacity must be calculated graphoanalytically [3], dividing the channel bandpass by a series of intervals with constant spectral density of noise output, which is possible due to the relative narrow-banded nature of the hydrophones and of hydroacoustic emitters.

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GEOPHYSICS, ASTRONOMY AND SPACE

UDC 534.24

FLUCTUATIONS OF THE ANGLE OF ARRIVAL OF A BEAM UPON REFLECTION FROM A STATISTICALLY UNEVEN SURFACE

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 26-27

[Article by V. M. Frolov]

[Text] Fluctuations of the angle of arrival of a beam to the detector are observed during experimental investigation of sound scattering on an agitated sea surface. This problem is considered theoretically in the given paper in beam approximation for sloping large-scale unevennesses and estimates of the possible fluctuations of angles are made by the derived formulas. It is assumed that there is no shading of some sections of the surface by others.

Let us use beam theory, assuming that reflection occurs according to laws of geometric acoustics from a plane tangent to the surface at the reflection point. Let us write the equation of an uneven surface in the form $z = \zeta = \zeta(x, y)$, where the random function ζ is the shifting of the surface with respect to the mean plane $z = 0$. Let the beam at given location of the source and detector in the plane xz at depths of z_0 and z, respectively, be specularly reflected at sliding angle ψ_0 at some point σ in the case of an even surface. One can compile some representation of the fluctuations of the direction of beam arrival to the detector upon reflection from an uneven surface, assuming that the reflecting surface, remaining locally flat, rotates around point σ by a random angle and is combined along the vertical by a random distance ζ, identifying these terms and combinations with random slopes and mixing of the agitated surface. This model permits one to find the angles of arrival of the beam at the observation point by the imaginary source method.

The following formulas are found on the assumption of the smallness of angles φ_x and φ_y between an uneven surface and axes x and y and also of the value $\varepsilon = \zeta/r$ (r is the distance between the imaginary source and the detector in the case of an even surface) and with regard to first- and second-order terms of smallness of these values for mean and mean square deviations $\delta\psi = \psi_0 - \psi$, $\delta\chi = \chi_0 - \chi$ and $\delta\vartheta = \vartheta_0 - \vartheta$ of angles ψ, χ and ϑ between the direction of arrival of the beam and axes x, y, z from undisturbed values ψ_0, $\chi_0 = \pi/2$ and $\vartheta_0 = \pi/2 - \psi_0$:

$$\langle \delta \psi \rangle \simeq 2\sigma^2 \frac{\sin 2\psi_0}{z^2} + \gamma_y^2 \frac{\eta}{(1+\eta)^2} \sin 2\psi_0, \tag{1}$$

$$\langle \delta \chi \rangle = 0, \tag{2}$$

$$\langle \delta \vartheta \rangle = -2\sigma^2 \frac{\sin 2\psi_0}{z^2} - 2\gamma_y^2 \frac{\operatorname{tg} \psi_0}{(1+\eta)^2} (1 + \eta \cos^2 \psi_0), \tag{3}$$

$$\langle \delta \psi^2 \rangle = \langle \delta \vartheta^2 \rangle = \sigma^2 \left(\frac{2 \cos \psi_0}{z}\right)^2 + \gamma_x^2 \left(\frac{2}{1+\eta}\right)^2, \tag{4}$$

$$\langle \delta \chi^2 \rangle = \gamma_y^2 \left(\frac{2 \sin \psi_0}{1+\eta}\right)^2. \tag{5}$$

Here $\eta = z/z_0$ and $\sigma = \sqrt{\langle \xi^2 \rangle}$ is the mean square height of the unevennesses and $\gamma_x = \sqrt{\langle \varphi_x^2 \rangle}$ and $\gamma_y = \sqrt{\langle \varphi_y^2 \rangle}$ are the mean square values of the fluctuations of angles between an uneven surface and planes yz and xz.

The following should be used at angles ψ_0 close to $\pi/2$ instead of formulas (3) and (4) to calculate $\langle \delta \vartheta \rangle$ and $\langle \delta \vartheta^2 \rangle$:

$$\langle \vartheta^2 \rangle = \vartheta_0^2 + \left(\frac{2}{1+\eta}\right)^2 (\gamma_x^2 \eta^2 + \gamma_y^2) .$$

As can be seen from formula (3), the mean value $\delta\vartheta$ is negative and, consequently, $\langle \vartheta \rangle$ is greater than ϑ_0. This means that the main sliding angle of the arriving beam with respect to the horizontal plane is less than the sliding angle upon reflection from an even surface.

The permissibility of replacing the section of uneven surface by a plane is limited by the requirement that the reflection point of the beam arriving at the detector lag behind the point of mirror reflection from the mean plane by a distance much less than the typical scale of correlation of surface roughness l. This requirement leads to the following restrictions:

$$\frac{\sigma}{l} \ll \left|\frac{1+\eta^2}{1-\eta}\right| \operatorname{tg} \psi_0 ; \quad \gamma_x \ll \frac{l(1+\eta)}{2z} \sin^2 \psi_0 ; \quad \gamma_y \ll \frac{l(1+\eta)}{2z} .$$

To estimate possible fluctuations of angles, let us consider the case when $\psi_0 = 75°$, $z_0 = z = 10$ m, $\sigma = 0.3$ m and $\gamma_x = \gamma_y = 6°$. The derived formulas then yield the values: $\langle \delta\vartheta \rangle = -1.3°$, $\sqrt{\langle \delta\vartheta^2 \rangle} \simeq 6°$ and $\sqrt{\langle \delta\chi^2 \rangle} = 5.5°$. The given example shows that fluctuations of the angles of arrival of the beam may be rather significant.

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GEOPHYSICS, ASTRONOMY AND SPACE

UDC 534.883.5

MODELS OF ECHO SIGNALS FROM OBJECTS OF COMPLEX SHAPE DURING PULSED EMISSION

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 29-32

[Article by G. D. Filin]

[Text] The concept of the equivalent scattering area S_e is used for the characteristic of the reflecting properties of sonar objects (0) (fish shoals, cliff projections, apparatus for underwater investigations, rescue apparatus and so on). Adequate description of the required class 0 should yield the probability description S_e (probability density $f(S_e)$) and the law of variation S_e in time (spectrum). The method of forming the models of 0 is the following. The dependence of S_e on the angle of approach $S_e(\beta)$ is initially determined. Description of the trajectory of relative motion permits one to establish the probability of realizing one or another angles of approach β and the dependence of β on time, which in combination with data on $S_e(\beta)$ makes it possible to determine $f(S_e)$ and the spectrum of S_e. Some aspects of constructing the models of 0 are considered in [2-13].

A general survey of papers on problems of analyzing the structure of echo-signals from bodies of different shape is presented in [12]. We felt that a more adequate diagram of analysis of the field structure of bodies of complex shape in the region of boundary and surface scattering is now the Freedman scheme [13].

According to [13], echo-signals are transmitted by individual "points" 0, in which the function $dS(r)/dr$ ($S(r)$ is the projection of that part of the scattering 0 to the reading plane which is found within the limits of (r_1, r); r_1 is the distance to the nearest point 0) or its derivatives undergo a final break. In this case

$$S_э = \left\{ \sum_{i=1}^{m} \sqrt{S_i} \exp\left[-j2k(r_i - r_1)\right] \right\}^2, \quad k = 2\pi/\lambda,$$

where $\sqrt{S_i} = 4\pi/\lambda \sum_{n=0}^{\infty} \mathcal{D}(i,n)/(jk)^n$; $\mathcal{D}(i,n) = S_-^{(n)}(r_i) - S_+^{(n)}(r_i)$ is the value of the jump of the n-th derivative of $S(r)$ at point r_i. Some characteristics of field formation under conditions of nonmonotonic emission were investigated in [10].

There is a number of 0 on individual sections of the surface of which $S^{(n)}(z)$ undergo an infinite break. The concept of an echo-signal is shown in [10] in this case on the example of a cylinder irradiated normal to the axis.

Four types of echo-signal envelopes occur:

1. When $|r_i - r_j| \gg c\tau_0/2$ (τ_0 is the length of the sounding signal), the echo-signal consists of individual pulses which repeat the shape of the transmission.

2. At $|r_i - r_j| \ll c\tau_0/2$, the field is formed by many components at each moment and, according to phase disorderliness, the shape of the echo-signal envelope is formed according to probability laws.

3. The shape of the echo-signal envelope is found according to the laws of vector addition under conditions of paragraph 2 where there are several overlapping pulses.

4. When $c\tau_0/2 \gg l_{max}$, the echo-signal consists of a central part and relatively short fronts. It essentially repeats the shape of the transmission, while the value of the envelope may be predicted if the values and phases of the components are known.

Excluding the case of a long transmission ($c\tau_0/2 \gg l_{max}$), there may be two types of phase structure of an echo-signal. The resulting phase is determined by laws of vector addition with a small number of simultaneously overlapping pulses. It is determined by probability laws with a large number of pulses.

Position O with respect to the observer (N) is characterized by the following parameters: α and β are heading angles N and O, Ω is the angular velocity of O during motion through the water basin, γ is the relative heading O and r is distance. The probability densities of values α and β are determined to a significant degree by the type of relative trajectory of motion O through the water basin, i.e., by the topology of the search plane. If the topology is point (random motion of O through the water basin), then $f(\alpha)$ and $f(\beta)$ are cosinusoidal in nature and $\overline{\Omega} \simeq 9$ deg/min; if it is linear (i.e., the trajectories of motion O are quasi-determined functions), which is natural when the direction of motion or the band γ is known, then $f(\alpha)$ and $f(\beta)$ are essentially uniformly in the permissible ranges of α and β and $\Omega \simeq 5$ deg/min. Rapid fluctuations caused by yawing of O and N are observed for relatively slow variations of β at velocities Ω. The distribution of β with normal yawing is frequently assumed. Normality is provided by the presence of the properties of symmetry and the independence of individual values of β.

In solving many problems related to construction of processing diagrams, information about the location of an echo-signal in the space of parameters is often significant, that is, at which levels, length, frequency and so on one may expect its appearance. Models of the fluctuations of the echo-signal envelope, realized under different conditions, are presented in [10].

The main causes of fluctuations of echo-signal length (τ_e) are ambiguity of the position of O by range (r) and depth (h) and also the random nature of the angle of approach β, so that the following relation occurs [1, 7, 11]:

$$\tau_{\text{э}} = \tau_0 + \frac{4}{3} \cdot \frac{r}{c_0} \Delta\alpha \cdot \alpha_0 + \frac{2l}{c} \cos\beta = \tau_0 + \tau(h, r) + \tau(\beta),$$

where τ_0 is the length of the sounding radiation, $\Delta\alpha$ is the aperture of the radiation pattern characteristic, α_0 is its slope and c_0 is the speed of sound at the depth of radiation. The probability density τ_e determined by the ambiguity of the angle of approach β is presented in [11]. With uniform distribution of O by h and with Rayleigh $R(\sigma_r^2)$ distribution of O by r:

$$f(\tau_{\text{э}}) = \frac{\sqrt{2}}{\Delta\alpha_0\, a\, \sigma_r} \Gamma\left(\frac{1}{2}, \frac{\tau_{\text{э}} - \tau_0 - \tau(\beta_0)}{2(\Delta\alpha_0)^2 a^2 \sigma_r^2}\right),$$

where $a = 4\Delta\alpha/3c_0$, $\Delta\alpha_0$ is the maximum slope of the receiving antenna and $\Gamma(\ldots, \ldots)$ is an incomplete gamma-function. The frequency of the sine-wave signal reflected by the i-th luminous point at the location of $w_i = w_0 + \Delta w_{to} + \Delta w_{gl} + \Delta w_0$ and is determined by the type of trajectory of relative motion $O(\Delta w_{to})$, the geometry of beam curvature in the medium (Δw_{gl}) and also by random displacement of the luminous point with respect to the center of mass (Δw_0). The indicated components are considered in [9].

The analytical problem of determining the spectral density of fluctuations S_e in the range of $(0, f_{max})$ is based on the assumption of the nature of probability distribution β during yawing. In the case of normal distribution law, the spectral density and correlation function of the reemission fluctuations are equal to $\sigma(f/f_{max}) = (1/2\pi)\exp(-f^2/2)$ and $R(\tau) = \exp(-\tau^2/2)$ and is determined by the value of realized during the process of operation.

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GEOPHYSICS, ASTRONOMY AND SPACE

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THE PROBLEM OF ANALYZING THE ENERGY ANOMALIES OF REVERBERATION FROM SOUND-SCATTERING LAYERS

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 45-48

[Article by V. N. Goncharov and V. V. Ol'shevskiy]

[Text] As is known (see, for example, [1-3]), sound scattering layers are encountered in almost the entire water basin of the world ocean. The scattering coefficients of these layers are so great that in most cases of practical interest, scattering from sound-scattering layers makes the main contribution to the actually observed reverberation. In this regard the problem of calculating the levels of reverberation from the sound-scattering layers is very timely.

The expression for the reverberation level (see [4]) has the form:

$$\mathcal{J}_{pH}(D)=\frac{P_A K_p}{16\pi^2}\sum_{i,j=1}^{n}\frac{\Delta V_{ij} f_i f_j \gamma_{pHij}}{D^4}\exp(-4\beta_{ij} z_i), \tag{1}$$

where P_A is the emitted acoustic power, K_p is the volumetric scattering coefficient in the layer and ΔV_{ij} is the elementary scattering volume determined as

$$\Delta V_{ij}=\begin{cases}\Delta V_i, & i=j,\\ \Delta V_i \cap \Delta V_j, & i\neq j,\end{cases}$$

and

$$\Delta V_i \cap \Delta V_j = \Delta V_j \cap \Delta V_i,$$

f_i and f_j are factors of focusing the corresponding beams (see below), $\gamma_{pm_{ij}}$ is the value of the scattering indicatrix of the scattering layer in a direction opposite to the direction of beam arrival, determined as

$$\gamma_{pMij} = \begin{cases} \gamma_{pMi}, & i=j, \\ \gamma_{pMi} + \gamma_{pMj}, & i \neq j, \end{cases}$$

r_{ij} is the acoustic length of beam travel, β_{ij} is the absorption coefficient for the corresponding beam and D is the horizontal distance to the center of gravity of the corresponding volume from the Z-axis.

Along with the level of reverberation from the layer, it is of interest to calculate the so-called reverberation anomaly, determined as

$$A_p(D) = 10 \lg \frac{J_{pM}}{J_{po}}, \tag{2}$$

where I_{po} is the reverberation intensity in an infinite medium without refraction with standard absorption and I_{pm} is the multibeam reverberation intensity in a layered-inhomogeneous medium with actual absorption.

The expression for the reverberation anomaly $A_p(D)$ has the form

$$A_p(D) = \frac{1}{\pi H_c T_{эф}} \sum^{n} \frac{\Delta V_{ij} f_i f_j \gamma_{pMij}}{D \gamma_{poij}} exp[-4(\beta_{ij} r_{ij} - \beta_o D)], \tag{3}$$

where H is the thickness of the layer, c is the speed of acoustic wave propagation in a homogeneous medium, T_{ef} is the effective length of the emitted pulse, β is the standard absorption coefficient in the medium without refraction and $\gamma_{po_{ij}}$ is the value of the scattering indicatrix for a medium without refraction. The values of ΔV, f, r and D are calculated by formulas of beam acoustics, while the remaining values are either system or hydrological characteristics.

Let us now judge some details of the adopted model. It must first be noted that the following assumptions are used in the calculations: the medium is layered-inhomogeneous, the focusing factor for one beam in the acoustics zone does not become infinite as follows from the nature of beam approximation, but is limited at a level of +25 dB, which agrees to a sufficient degree with experimental data, and finally so-called "bottom" beams, i.e., beams reflected from the bottom, are not taken into account in the calculation.

Reception and emission for most types of hydrologies are limited in the calculation to a range of ±15° with respect to the horizontal, which is determined by the condition of the absence of "bottom" beams. The receiver and emitter are located at the same point and are nondirectional.

Let us now consider some characteristics of the process of calculation by formulas (1) and (3). One of the most significant elements of calculation is division of the angular range indicated above into angles corresponding

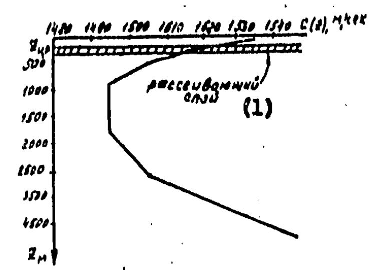

Figure 1. Speed of Sound Profile c(z), Location of Scattering Layer (Dashed Region) and Position of Antennas -- Point Z_i

KEY: 1. Scattering layer

to individual beam congruences. As is known, congruence or a bundle of beams is a set of beams having identical number of reflections from the bottom and surface and also an identical number of turns. Moreover, all beams of this set should pass through the same layers.

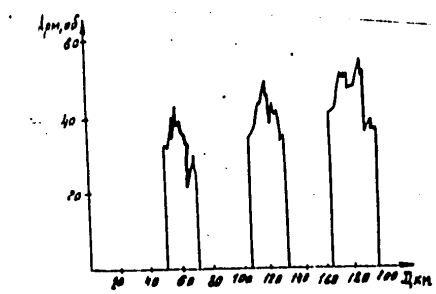

Figure 2. Multibeam Reverberation Anomalies for Scattering Layer 100 m Thick Whose Lower Boundary is Located at a Depth of 300 m: a -- first far zone of acoustic illumination; b -- second zone; c -- third zone

Beams having the indicated properties have a continuous (outside the caustic regions and boundaries) and an almost plane wave front, which permits consideration of them as an individual beam with some mean intensity (along the front). Moreover, division into bundles considerably simplifies the process of calculations, making it unnecessary to take into account too large a number of beams.

A program in Fortran language was compiled for calculation by formulas (1) and (3) which permits calculation of the reverberation level $I_p(D)$ and its anomaly $A_p(D)$ from sound-scattering layers for distances up to 200 km.

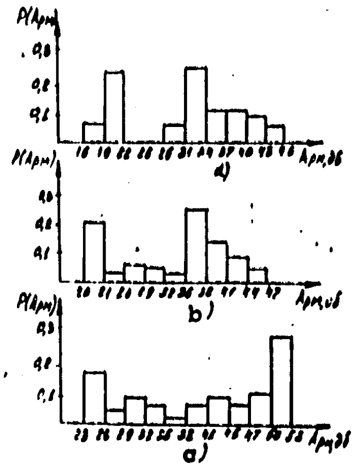

Figure 3. Histograms of Distribution of Multibeam Reverberation Anomalies $A_{pm}(D)$ in Three Far Zones of Acoustic Illumination: a -- first zone; b -- second zone; c -- third zone

The anomalies of reverberation from a sound-scattering layer for a specific hydrology (see Figure 1), characterized by the presence of an underwater sound channel, were calculated as an example by using the compiled program. This hydrology is typical for many regions of the world ocean in which sound-scattering layers are observed. The parameters of the sound-scattering layer are as follows (see Figure 1): depth of the upper boundary -- 200 m, depth of the lower boundary -- 300 m, the emitter and receiver were located at the same point at a depth of 150 m.

Reverberation anomalies $A_p(D)$ were calculated (see Figure 2) for the first three far zones of acoustic illumination. Moreover, histograms of the distribution of reverberation anomalies by the three far zones of acoustic illumination (see Figure 3), which correspond to the considered case, were constructed. As follows from Figures 2 and 3, the values of the reverberation anomaly may reach significant values -- up to +55 dB. The zonal structure of long-range reverberation is clearly manifested and in this case the width of the zones and the value of the anomaly increase with an increase of the number of the zone.

In conclusion the authors take this opportunity to express their gratitude to colleagues of the Acoustics Institute Professor Yu. M. Sukharevskiy, junior research associate V. Yu. Postnikova and also L. S. Rayskaya for useful discussions of the materials of the paper.

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GEOPHYSICS, ASTRONOMY AND SPACE

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SPACE-TIME CORRELATION OF THE REVERBERATION NOISE FIELD

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[Article by V. V. Krizhanovskiy and S. V. Pasechnyy]

[Text] The characteristics of the space-time reverberation correlation function (KF) based on the wave approach are investigated in the paper. This consideration permits one to take into account rather fully the effect of individual factors and systems characteristics on the KF structure. Scatterers are characterized by KF (or scattering indicatrix) of arbitrary form. The correlation structure of the far volumetric reverberation field is investigated within the framework of the small perturbation method with arbitrary arrangement of the source and detector with respect to each other. The medium is assumed to be random-inhomogeneous in the sense of [2].

Reverberation in an Unlimited Random-Inhomogeneous Medium

The field velocity potential of a signal multiply scattered in the Fraunhofer zone with respect to the emitter, with regard to [2] and [3], may be written in the form:

$$\varphi_1(\vec{R}_n;t) = -\frac{K_0^2}{16\pi^2}\int\limits_{V_p}\frac{\mu(R_p,t-\frac{R_{pn}}{c})}{R_{up}R_{pn}}S(t-\frac{R_{up}+R_{pn}}{c})F_u(\frac{\omega_0}{c}\vec{m})d\vec{R}_p, \quad (1)$$

where $\mu(.,.)$ are fluctuations of the refractive index; $S(\cdot)$ is an arbitrary narrowband signal; $F_u(\cdot)$ is the characteristic of emitter directionality at the medium frequency of the signal spectrum; $K_0 = \omega_0/c$ is a wave number; R_{ir} and R_{rp} are the distance from the point of the scattering volume $\vec{R}_r$ to the center of the emitter $\vec{R}_{i_0}$ and the reception point $\vec{R}_p$, respectively; and $\vec{m}$ is the unit vector from the center of the emitter to the point of the scattering volume V_r.

Multiplying this expression by a value shifted in time and space, averaging by the set, converting to relative coordinates $\vec{\rho} = \vec{R}_r' - \vec{R}_r''$ and $\vec{D} = \vec{R}_p' - \vec{R}_p''$ and to the coordinates of the center of gravity $\vec{x} = \vec{R}_r' + \vec{R}_r''/2$ and

$\vec{y} = (\vec{R}_p' + \vec{R}_p'')/2$ and assuming that the receiving base D is located in the Fraunhofer zone with respect to the scattering region, we find the space-time KF of the following form for a homogeneous and stationary field of scatterers located in the Fraunhofer zone with respect to the emitter and the reception point:

$$B(t,\tau,\vec{\mathcal{D}},\vec{R}_{u0},y) = \frac{K_0^4}{256\pi^4} \iint_{V_\rho V_x} B_\mu(\vec{\rho},\tau) S\left(t - \frac{R_{up}+R_{pn}}{c} - \frac{\vec{\kappa}\vec{\rho}+\vec{n}\vec{\mathcal{D}}}{2c}\right) \times S^*\left(t-\tau-\frac{R_{up}+R_{pn}}{c}+\frac{\vec{\kappa}\vec{\rho}+\vec{n}\vec{\mathcal{D}}}{2c}\right)\left|F_u(\vec{m})\right|^2 \frac{d\vec{x}\,d\vec{\rho}}{R_{up}^2 R_{pn}^2}, \quad (2)$$

where $B_\mu(\vec{\rho},\tau) = \langle \mu(\vec{x}+\frac{\vec{\rho}}{2},t)\mu(\vec{x}-\frac{\vec{\rho}}{2},t-\tau)\rangle$; $\tau = t'-t''$, $\langle\mu\rangle = 0$, $\vec{m} - \vec{n} = \vec{k}$ is the scattering vector, $\vec{n}$ is the unit vector from some point of the volume of scatterers $\vec{x}$ to the center of the receiving base $\vec{y}$.

Following [3], let us turn in expression (2) from KF of $B_\mu(\vec{\rho},\tau)$ to the averaged spatial spectral density of the output of inhomogeneities $\bar{\Phi}_\mu(\omega_0/c, \vec{k}, \tau)$, using the relation:

$$\bar{\Phi}_\mu\left(\frac{\omega_0}{c}\vec{\kappa},\tau\right) = \frac{1}{(2\pi)^3}\int_{V_\rho} B_\mu(\vec{\rho},\tau) e^{i\kappa_0\vec{\kappa}\vec{\rho}}\,d\vec{\rho}. \quad (3)$$

As a result, assuming that $\bar{\Phi}_\mu(.,.)$ is unchanged in the frequency band of the signal, we find

$$B(t,\tau,\vec{\mathcal{D}},\vec{R}_{u0},\vec{y}) = \frac{K_0^4}{32\pi}\int_{V_\rho} \bar{\Phi}_\mu\left(\frac{\omega_0}{c}\vec{\kappa},\tau\right) S\left(t-\frac{R_{up}+R_{pn}}{c}-\frac{\vec{n}\vec{\mathcal{D}}}{2c}\right) \times S^*\left(t-\tau-\frac{R_{up}+R_{pn}}{c}+\frac{\vec{n}\vec{\mathcal{D}}}{2c}\right)\left|F_u(\vec{m})\right|^2 \frac{d\vec{x}}{R_{up}^2 R_{pn}^2}. \quad (4)$$

Expression (4) permits one to find the indicatrix of the reverberation field with regard to the direction of signal scattering by inhomogeneities of the medium.

Let us consider the characteristics of the effect of individual components contained in (4) on space-time KF on a number of examples. Let us assume that the signal is "short" so that the effect of attenuation factors may be disregarded within its length. We will also subsequently disregard time variation of the KF of random inhomogeneities on the interval of signal length and variation of the signal generatrix at distances on the order of the maximum dimensions of the inhomogeneities ρ_{max} and of the receiving base D_{max}.

In the special case of a nondirectional source and one combined with the detector and δ-correlated (or small-scale) scatterers which fill the entire volume, from (4) we find (as in [1, 4]):

$$B(t,\tau,\mathcal{D}) = \frac{\sigma_0}{2\pi c t^2} K_s(\tau) \frac{\sin K_0 \mathcal{D}}{K_0 \mathcal{D}}, \qquad (5)$$

where

$$K_s(\tau) = \int_{\tau}^{T} S(t) S^*(t-\tau) dt; \; \sigma_0 = \frac{\pi}{2} K_0^4 \Phi_\mu (2 \frac{\omega_0}{c} \vec{m}).$$

It is obvious from (5) that the reverberation field is homogeneous and isotropic in this case and its KF is divided into space and time functions. The dependence on time is contained only in dispersion, which indicates the possibility of making it stationary.

Let us consider the effect of the geometry of the scatterer volume with the assumptions indicated above. If the scatterers are concentrated in a thin horizontal layer of thickness h and if the effect of the final dimensions of illumination of the region on phase variation of the scattered signals added at the reception point may be disregarded, then:

$$B(t,\tau,\vec{\mathcal{D}},z_p) \approx \frac{\sigma_0 h}{2\pi c^2 t^2} K_s(\tau) J_0\left[K_0 d \sqrt{1 - \frac{4 z_p^2}{c^2 t^2}}\right] e^{i K_0 \frac{2 z_p}{ct} z_D}, \qquad (6)$$

where z_r is the height of the center of the scattering layer above the level of the emission-reception point and $\vec{D} = \{d \cos \varphi, d \sin \varphi, z_D\}$. As we can see, restriction of the dimensions of the scattering volume in the case of noncoincidence of the geometry of the illuminated region to that of this volume, leads to the appearance of anisotropy of the reverberation field. Let us also indicate that the field becomes inhomogeneous and transiently bound through the space with the source and detector located outside the scattering layer.

Let us determine the role of the anisotropy of scatterers under the conditions indicated in the first example. Being given the correlation function of the scatterers in the form:

$$B(\vec{\rho}) = \sigma_\mu^2 \exp\left\{-\frac{\rho x^2 + \rho y^2}{d^2} - \frac{\rho_z^2}{p^2}\right\}.$$

or the spatial spectral power density corresponding to it

$$\Phi_\mu(\vec{\ae}) = \frac{\sigma_\mu^2 a^2 \rho}{8\pi^{3/2}} \exp\left\{-\frac{a^2(\ae_x^2+\ae_y^2)}{4} - \frac{\rho^2 \ae_z^2}{4}\right\},$$

we find the model of the anisotropic field of scatterers in the vertical plane. With regard to this, the KF of the reverberation field in the vertical direction assumes the form:

$$B_\theta(t,\tau,\mathcal{D}) = \frac{\sigma_0 e^{-\kappa_0^2 a^2}}{8\pi c t^2 \kappa_0 \sqrt{\rho^2-a^2}} \kappa_s(\tau) e^{-\frac{\mathcal{D}^2}{4(\rho^2-a^2)}} \times$$
$$\times\left[\Phi\left(-\frac{i\mathcal{D}}{2\sqrt{\rho^2-a^2}} + \kappa_0\sqrt{\rho^2-a^2}\right) + \Phi\left(\frac{i\mathcal{D}}{2\sqrt{\rho^2-a^2}} + \kappa_0\sqrt{\rho^2-a^2}\right)\right], \quad (7)$$

where $\Phi(\cdot)$ is the probability integral and $\sigma_0 = (\sigma_\mu^2 a^2 p k_0^4)/16$. For correlation in the horizontal direction we have:

$$B_r(t,\tau,\mathcal{D}) = \frac{\sigma_0 e^{-\kappa_0^2 a^2}}{2\pi c t^2} \kappa_s(\tau) \int_0^1 e^{-\kappa_0^2(\rho^2-a^2)x^2} J_0\left[\kappa_0 \mathcal{D}\sqrt{1-x^2}\right] dx. \quad (8)$$

As r = a, expressions (7) and (8) coincide with (5) with accuracy up to amplitude factors. This indicates that with combined emission-reception the structure of spatial correlation does not depend on the scales of the scatterers if they are isotropic, but a difference is observed only at levels equivalent to variation of the effective back-scattering cross-section. In the general case the anisotropy of the scattering field leads to anisotropy of the reverberation field. We also note that if the anisotropy of the field is sharply marked ($r \gg a$ and $k_0 r \gg 1$), then expression (7) is transformed to the form:

$$B_\theta(t,\tau,\mathcal{D}) \approx \frac{\sigma_0 e^{-\kappa_0^2 a^2} \kappa_s(\tau)}{4\pi c t^2 \kappa_0 \sqrt{\rho^2-a^2}} e^{-\frac{\mathcal{D}^2}{4(\rho^2-a^2)}}.$$

Thus, one can talk about a "transfer" of the correlation function in directions parallel to the maximum spatial scale of KF of scatterers. In this case the radius of spatial correlation increases twofold.

Correlation in the horizontal direction at $p \gg a$ and $k_0 p \gg 1$ has the form:

$$B_r(t,\tau,\mathcal{D}) \approx \frac{\sigma_0 e^{-\kappa_0^2 a^2}}{4\pi c t^2 \kappa_0 \sqrt{\rho^2-a^2}} \kappa_s(\tau) J_0(\kappa_0 \mathcal{D}).$$

This expression coincides in structure with the correlation function in layer (6) (at $z_p = 0$). Thus, the marked anisotropy of the scattering field in the vertical direction leads to concentration of the main energy of the scattered signals in horizontal directions, which is equivalent to shifting of the receiving-emitting system to the horizontal layer. The role of the remaining part of the scattering volume is reduced to variation of energy factors.

If the emission and receiving points coincide, the directional properties of the scatterers and the emitter, according to (4), may affect the structure of the KF in an identical manner. Specifically, if the normalized characteristic of the linear vertical radiation pattern of the emitter antenna is given in the form of $F(\vec{m}) = \exp\left\{-k_0^2 p^2 \cos^2\theta\right\}$, then we will find an anisotropic reverberation field for small-scale scatterers, coinciding in structure to the previous case (at $a \to 0$).

Let us turn to analysis of the effect of separation of the receiving and emission points. This problem is outlined in [5] for a discrete model of scatterers, where it is shown that separation leads to the appearance of transient nature of the indicatrix of the reverberation field. The experimental results of [6] indicate the presence of inhomogeneities whose scales satisfy the relation $k_0 a > 1$, for typical echo-location frequencies. These inhomogeneities are characterized by scattering indicatrices extending in the "forward scattering" direction. Therefore, during separated emission-reception, one must additionally take into account the effect of the directional properties of scatterers on the indicatrix of the reverberation field. Expression (4) generalizes the result of [5] in this layout.

Let us consider as an example a volumetric reverberation field with horizontal separation of the emission and reception points and of the task of the directional properties of the source and scatterers in the form:

$$F_u(\theta_u) = \begin{cases} 1, \text{ npu } \theta_u \in [-\theta_{u0}, \theta_{u0}]; \\ \delta, \text{ npu } \theta_u \in [-\theta_{u0}, \theta_{u0}], \end{cases}$$

$$\bar{\Phi}_\mu(\varkappa) = \exp\left\{-\frac{a^2\varkappa^2}{4}\right\}, \text{ ege } \varkappa = \kappa_0|\vec{R}|,$$

θ_i is an angle read from the horizontal axis which combines the source and detector given in the coordinate system of the source.

The indicatrix of the reverberation field is determined in this case by the expression:

$$C(t,\theta_n) = M_0(t) M_1(t,\theta_n) \bar{\Phi}_\mu(\varkappa) F_u(\theta_n), \tag{9}$$

where $M_0(t) = \frac{2e^{-\beta ct}}{(ct)^2}$ is the attenuation factor; $M_1(t,\theta_n) = \frac{1}{1+2\varepsilon_t \cos\theta_n + \varepsilon_t^2}$ is a function which characterizes the directionality of the reverberation field, which appears due to separation of the source and detector; let us call it the geometric indicatrix;

$$\theta_n = \arccos\left\{\frac{(1+\varepsilon_t^2)\cos\theta_u - 2\varepsilon_t}{1+\varepsilon_t^2 - 2\varepsilon_t \cos\theta_u}\right\}$$

is the angle given in the coordinate system of the detector;

$$æ = K_0\left[\frac{2(1+\varepsilon_t \cos\theta_n)}{(1+2\varepsilon_t \cos\theta_n + \varepsilon_t^2)^{1/2}}\right]; \quad \varepsilon_t = \frac{R_{io}}{ct},$$

and R_{io} is the distance between the source and detector.

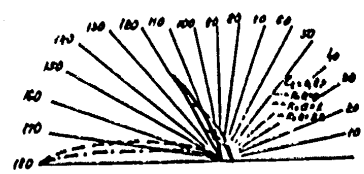

Figure 1

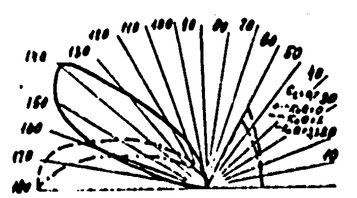

Figure 2

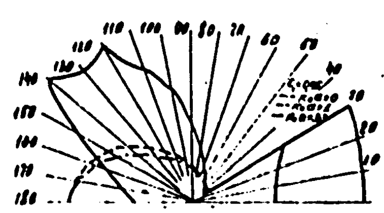

Figure 3

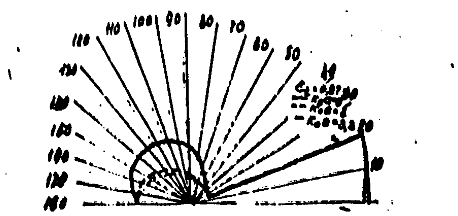

Figure 4

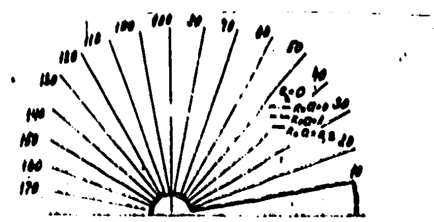

Figure 5

Graphs which illustrate the dependence of the indicatrix (9) on the time and scales of inhomogeneities provided that normalization is maximum (it was assumed in the calculations that $\delta^2 = 10^{-1}$ and $\theta_{1o} = 10°$) are presented in Figures 1-5.

It is obvious from the graphs that even at $k_o a = 3.3$, the direction of scattering has a significant effect on the direction of the reverberation field. Specifically during the initial moments of time reverberation related to emission of the aureole of the characteristic in directions corresponding to the minimum value of the scattering vector $\vec{K}$ whereas the indicatrix of the reverberation field is extended in a direction toward the source without regard to the direction of the scatterers, makes a significant contribution to the total noise output. Variation of the direction of the reverberation field in time leads to transient linking of the reverberation field in space. We also note that if variations of the direction of scatterers within the range of projection of the illuminated region onto the direction to the observation point cannot be disregarded, the autocorrelation function of the reverberation field also becomes transient. Besides this, as noted in [5], the transient nature of autocorrelation may also be caused by the effect of the direction of the source and of the geometric indicatrix.

Volumetric reverberation in a randomly inhomogeneous waveguide. Using the concept of imaginary sources [8], one can show that the space-time reverberation field will be described by superposition of the terms of form (1), which describes the propagation of signals along different trajectories appearing as a result of taking into account reflections of the incident and scattered signals from the surface and bottom.

Analysis shows that in the general case the presence of reflecting boundaries leads to the appearance of complex interference of the KF. If the emission and receiving points consisting of components which comprise the KF are combined, one can determine the subsequence which in the range of ($t \to \infty$, $\beta \to 0$, $V \to 1$; V is the coefficient of reflection from the boundaries) will yield a KF as in an unlimited medium provided that the scattering field is homogeneous and isotropic within the entire waveguide, the source is nondirectional and the boundaries are absolutely reflecting. Along with the components indicated above, the KF contains components which correspond to correlation of signals passing along different trajectories. These components characterize the correlation of signals arriving at the reception point from different directions. Their contribution to the total KF will be significant if the sources are slightly directional and the difference of the distances to the scattering point for the correlated waves are not reflected by their trajectory in the time correlation. Let us also indicate that the effect of separation should be manifested for these components despite the fact that the detector and emitter are combined. The latter becomes understandable if one takes into account that the region from which scattered signals may arrive simultaneously is an ellipsoid. The condition of the absence of correlation between different components of the spatial spectrum of scatterers for homogeneous and steady scattering field should be fulfilled for the remaining components of the KF. Correlation of signals that have covered different paths along different trajectories corresponds to these components in the general case and it may occur even with time shifts exceeding the signal length. The following factors limit the number of terms in the KF in practice: the final signal length, limited dimensions of the scattering zone, diminution of the time correlation of scatterers, directional properties of the source and scatterers and absorption on the boundaries and in the medium.

Space-time correlation of the reverberation field in a layered-inhomogeneous waveguide. When investigating space-time KF, the reverberation noise field at great distances from the scattering zone must take into account the layered inhomogeneity of the waveguide. The results for this case were found by using WKB representation of the field of incident and scattered waves [8]. It was assumed in this case that scattering of waves propagated along beams also occurs the same as for plane waves of the same directions in the scattering zone. With regard to [8] and [9] and provided that the acoustic field hardly varies within the range of scales of correlation of inhomogeneities, the expression for KF may be written in the form:

$$B^{(\tau,\nu)(\lambda,\beta)}_{(e,j)(s,\rho)}(t,\tau,\vec{y},\vec{D},\vec{R}_n)=\frac{\omega_0^4}{8\cdot 8\pi C_0^4}\int\limits_{V_\rho}\bar{\Phi}_\mu\left(\frac{\omega_0}{C_0}\vec{K}^{(\tau,\nu)(\lambda,\beta)}_{(e,j)(s,\rho)},\ \tau-\frac{W_{s\rho_0}-W_{\lambda\beta_0}}{C_0}\right)\times$$

$$\times S\left(t-\frac{W_{ej_0}+W_{s\rho_0}}{C_0}\right)S^*\left(t-\tau-\frac{W_{\tau\nu_0}+W_{\lambda\beta_0}}{C_0}\right)\frac{F_{ej_0}F^*_{\tau\nu_0}}{n_{\tau_0}n_{e n_0}}\ \frac{A_{ej}A_{s\rho}A_{\tau\nu}A_{\lambda\beta}}{R_{ej}R_{s\rho}R_{\tau\nu}R_{\lambda\beta}}\,d\vec{\xi}$$

where

$$A_{ej} = \frac{R^2_{ej}}{\hat{R}^2_{ej}} V^2_{ej}; \quad \hat{R}_{ej} = \sqrt{K_{z_u} K_{z_o}} \sqrt{|\alpha_{ej} \beta_{ej} - \gamma^2_{ej}|} (n_{z_o})^{-1/2};$$

$$\alpha_{ej} = \frac{\partial^2 f_{ej}}{\partial K_x^2}\bigg|_{\vec{K}_{10}}; \quad \beta_{ej} = \frac{\partial^2 f_{ej}}{\partial K_y^2}\bigg|_{\vec{K}_{10}}; \quad \gamma_{ej} = \frac{\partial^2 f_{ej}}{\partial K_x \partial K_y}\bigg|_{\vec{K}_{10}}; \quad \vec{K} = \{\vec{K}_\perp, K_z\}.$$

f_{ej} is a function which characterizes the phase components of WKB-representation of the field; R_{ej} is the distance along the corresponding trajectory in a homogeneous medium with absolutely reflecting boundaries; C_0 is the speed of sound at a typical level; $W_{ej}(S, P)/C_0$ is the travel time of the incident (scattered) signal along the trajectory; F_{ej} is a function which characterizes the direction of the source in a smoothly homogeneous medium: the subscript "o" indicates the fact that the values of the functions are taken at steady points $\vec{k}_{10}$ or $\vec{Z}_{10}$, which are solutions of a system of equations (for given coordinates $\vec{R}_i$, $\vec{R}_r$ and $\vec{R}_p$) and characterize the incident and scattered wave

$$\frac{\partial f_{ej}}{\partial K_i} = 0; \quad \frac{\partial f_{sp}}{\partial \varkappa_i} = 0; \quad (i = 1, 2), (K_1 = K_x; K_2 = K_y, \varkappa_1 = \varkappa_x; \varkappa_2 = \varkappa_y).$$

Expression (10) is similar in structure to the expression for KF in a randomly inhomogeneous waveguide. However, besides the factors considered previously, the KF of the reverberation field in a layered-inhomogeneous waveguide may be affected by the inhomogeneities of field amplitude related to the anomaly of signal propagation characterized by the beam focusing factor.

Conclusions

1. Expressions are found in approximation of the small-perturbation method for space-time KF of a multiply scattered reverberation noise field with regard to the system characteristics for three models of a hydroacoustic channel: unlimited randomly inhomogeneous medium, randomly-inhomogeneous waveguide and layered inhomogeneous waveguide. In the latter case the result is found by using WKB-approximation.

2. The effect of individual factors on the structure of the KF is analyzed. Specifically, it is shown that:

-- localication of the scattered signal power in separate directions, determined by the geometry of the scattering volume, the final length of the signal or the direction of the properties of the source and scatterers, leads to the appearance of anisotropy of the reverberation field. For example, the ratio of the correlation radii in the vertical and horizontal directions $\rightarrow \infty$ at $\varepsilon z_r/ct \rightarrow 0$ with scattering from a thin horizontal layer;

-- the anisotropy of inhomogeneities leads to the appearance of the anisotropy of the reverberation field; restoration of the KF of large-scale scatterers in directions parallel to the maximum scale of the KF of scatterers is possible with marked anisotropy of the scattering field and the correlation radius increases twofold in this case; the structure of the KF reverberation field for isotropic large-scale inhomogeneities coincides with accuracy up to energy factors to the structure of the KF for small-scale inhomogeneities with combination of the emission and receiving points;

-- taking the direction of signal scattering by random inhomogeneities with separated emission-reception into account is characterized by a significant contribution of reverberation determined by emission of the aureole of the emitter radiation pattern in directions corresponding to small values of the scattering vector to the total output of reverberation noise at the reception point, specifically, this effect is significant at $k_0a > 3$ and $\mathcal{E}_t > 0.27$ for the Gaussian model of the KF of scatterers;

-- the presence of reflecting boundaries leads to the appearance of interference of the KF of multibeam reverberation fields; the components of the resulting KF may be divided into two main groups: the first group contains those components for which the trajectories of the correlated waves coincide in the forward and reverse directions and the second group includes those for which the trajectories are different; scattered fields in which the correlation of waves arriving from different directions occurs (even for stationary and homogeneous scattering fields) correspond to the second group.

We note that all the conclusions with respect to the KF structure of the reverberation field and its characteristics, presented in this paper for the case of scattering on the inhomogeneities of the refractive index, also retains its force for first-order scattering fields of a different nature (specifically, caused by one-time scattering on the ZRS and on surface bubbles).

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6521
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GEOPHYSICS, ASTRONOMY AND SPACE

UDC 534.843.242:681.883.67

SPATIAL CORRELATION OF REVERBERATION WITH REGARD TO ACOUSTIC ANTENNA RADIATION PATTERNS

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 60-61

[Article by T. A. Moroz and L. M. Chibisova]

[Text] The effect of the dimensions and configuration of acoustic detectors and emitter on the spatial correlation of long-range volumetric and surface reverberation in study of narrowband pulsed signals and with separated reception along the wave front is analyzed in the paper. The coefficient of spatial correlation of reverberation was determined as

$$R_{np}(r)=\frac{\int d\vec{R}\cos\frac{k\vec{R}\vec{r}}{R}\varphi_u^2\left(\frac{\vec{R}}{R}\right)\varphi_n^2\left(\frac{\vec{R}}{R}\right)}{\int d\vec{R}\;\varphi_u^2\left(\frac{\vec{R}}{R}\right)\varphi_n^2\left(\frac{\vec{R}}{R}\right)}, \tag{1}$$

where $R = |\vec{R}|$ is the distance to the scattering volume, $k = 2\pi/\lambda$ is the wave number, λ is the length of the emitted wave, $\varphi_{i,p}$ are the radiation patterns of the emitter and detectors and $r = |\vec{r}|$ is the distance between detectors.

Nondirectional emission and reception ($\varphi_i = 1$ and $\varphi_p = 1$) are usually considered for simplicity; then for volumetric reverberation [1, 2] we have

$$R_{np}(r)=\frac{\sin kr}{kr}; \tag{2}$$

for surface reverberation or reverberation from a thin layer [1, 3] we have:

$$R_{np}(r)\approx J_0(kr), \tag{3}$$

where I_0 is a zero-order Bessel function of the real argument.

Calculations of $R_{pr}(r/\lambda)$ were made in the given paper on the BESM-6 computer with regard to specific acoustic antenna radiation patterns of several types.

Antennas were considered in the form of oscillating spheres, groups of two point antennas, discrete linear groups of (n + 1) point antennas, continuous linear groups with length l, continuous annular groups, round piston diaphragms and also antennas in the form of rectangular piston diaphragms.

Analysis showed that the absolute value of the spatial correlation of reverberation decreases rapidly with an increase of antenna dimensions, i.e., the increase of the direction caused by the increase of antenna dimensions leads to limitation of the total scattering zone overlapped by their patterns, while the interval of spatial correlation of reverberation increases significantly.

When using solid (continuous) antennas, fluctuations of the coefficient of spatial correlation, inherent to nondirectional (point) antennas and slightly directional antennas of small dimensions (antenna dimension of $d \leqslant 0.5\text{-}0.7\lambda$) are rapidly damped with an increase of antenna dimensions.

When using acoustic antennas in the form of small-element discrete groups, fluctuating dependence of the spatial correlation coefficient on the ratio r/λ with some elements of periodicity are typical. The intensity of these fluctuations are significantly dependent on the range of ratios d/λ. "Resonance" bursts of the spatial correlation coefficient in regions of $r/\lambda \approx d/\lambda$ are observed.

Multielement ($n \geqslant 5$) discrete antennas yield spatial reverberation correlation of the same type as similar continuous antennas.

The range of spatial correlation of long-range surface reverberation and reverberation from a thin layer with directional emission-reception is somewhat wider than the range of spatial correlation of long-range volumetric reverberation.

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6521
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GEOPHYSICS, ASTRONOMY AND SPACE

UDC 621.391.16

ESTIMATING LOSSES OF NOISE STABILITY OF SOME CROSS-CORRELATION PROCESSING CIRCUITS

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 66-69

[Article by B. M. Golubev, V. A. Koptyayev and A. V. Kotkin]

[Text] The presently used polarity coincidence circuits for analog, digital-analog and digital correlators are usually constructed on the basis of an algorithm which may be represented in general form in the following manner:

$$y(t,t_з)=\int_{-\infty}^{\infty}h(t-t_1)\,x(t_1)\,x_0(t_1-t_з)\,dt_1,$$

where x(t) is the input process, $x_0(t)$ is the reference signal, h(t) is the pulsed response of the integrator, t_z is the signal delay time and y(t) is the correlator output effect. These cross-correlation processing circuits can be called signal-channel; their essential feature is the presence of high-frequency components of the multiplying process at the output which affect the correlator output effect with real integrator characteristics.

The effectiveness of signal processing may be improved if high-frequency components are suppressed directly at the integrator input. This signal processing may be accomplished by using systems which utilize quadrature components of the input process and of the reference signal, i.e., by two-channel correlation systems which are constructed on the basis of the algorithm [1]

$$y(t,t_з)=\frac{1}{2}\int_{-\infty}^{\infty}h(t-t_1)\left[x(t_1)\,x_0(t_1-t_з)+\hat{x}(t_1)\,\hat{x}_0(t_1-t_з)\right]dt_1,$$

where $\hat{x}(t)$ and $\hat{x}_0(t)$ are the Hilbert transforms of signals x(t) and $x_0(t)$, respectively.

Mathematical analysis of the noise resistance of the considered correlator circuits shows that the analog correlator, constructed by the algorithm of two-channel signal processing, has the greatest noise resistance among them. Noise resistance losses p (by the output signal/noise ratio) of a single-channel analog correlator and of two- and single-channel digital-analog correlators may be characterized by positive coefficients of γ, i.e., $p = \gamma$, compared to a two-channel analog correlator. The considered loss coefficients depend on $a = f_a/\Delta f$ -- the quantization parameter -- and $b = 2f_0/\Delta f$ -- the narrow-bandedness parameter (f_a is the quantization frequency and f_0 is the central or intermediate frequency of the signal spectrum).

It follows from the results of calculations that the loss coefficient of a single-channel analog correlator has a negligibly small value even with wideband signals ($b \simeq 1$), i.e., single- and two-channel analog correlator circuits are essentially equivalent.

The losses in a two-channel digital-analog correlator do not depend on the narrow-bandedness parameter b and consequently the required value of the output effect in this circuit may be provided independently of the intermediate signal frequency. Moreover, the very high efficiency of this system, i.e., the proximity of the loss value to zero, is achieved at $a \gg 1$ ($f_a \gg \Delta f$), which agrees with the theorem on quantification of narrow-band processes using the Hilbert transform [2].

The dependence of the loss coefficient of a single-channel digital-analog correlator on the quantification parameter a and parameter b/a is presented in Figure 1. The most significant feature of this dependence is the periodicity by variable b/a with a period equal to unity. A convenient feature of representing these functions for practical application is the cross-section of these surfaces by horizontal planes corresponding to different loss levels. A set of lines of equal losses for a single-channel digital-analog correlator circuit is presented in Figure 2.

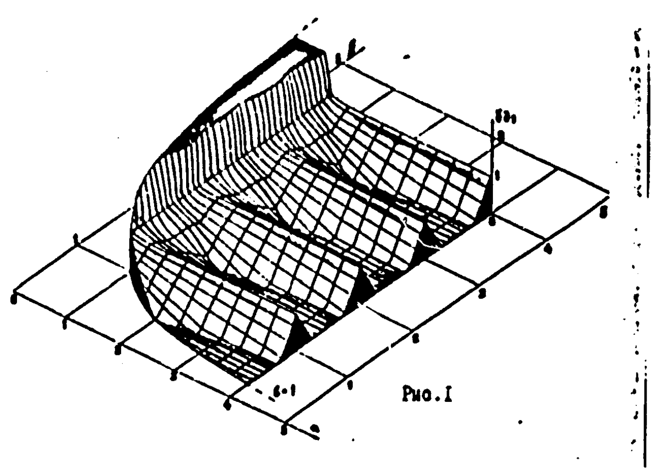

Figure 1

The loss functions in single- and two-channel polarity coincidence digital correlators (Figure 3, a and b, respectively) are presented in a similar manner. These functions also reflect the effects related to the presence of high-frequency components of the process at the integrator output. In this case their effect is also appreciable in a two-channel circuit sir total compensation of the high-frequency components does not occur in it due to signal limitation.

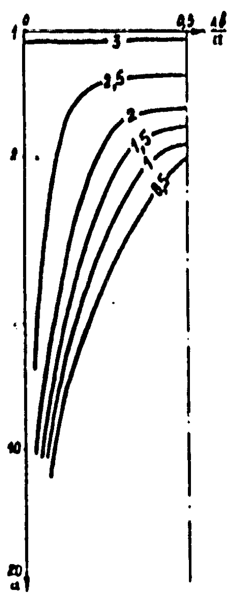

Figure 2

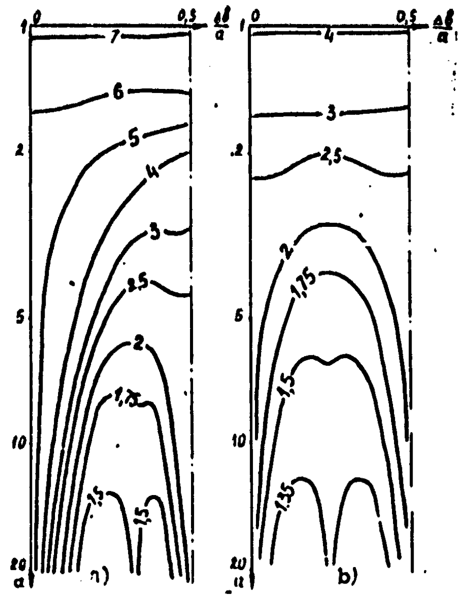

Figure 3

Comparison of the results of calculations to each other shows that the efficiency of signal processing by two-channel correlator circuits compared to single-channel circuits is considerably improved when using time quantification of the input processes and reference signals. An advantage in the quantification parameter, which may be selected in two-channel circuits at least half as much as in single-channel circuits, is achieved with equal losses in the output effect.

A number of experiments whose results are in good agreement with the main theoretical functions was carried out to check the conclusions reached.

The advantages of the values of storage capacity and speed of these devices required for this processing were calculated as a function of the relative Doppler shift of the received signal ($|\Delta f_g|/\Delta f$) to estimate the efficiency of two-channel correlators compared to single-channel correlators in signal processing in some range of their Doppler shifts using digital time-compression

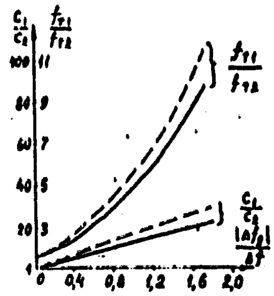

Figure 4. c -- storage capacity; f_t -- clock frequency

devices of input processes and reference signals [3]. The curves in Figure 4 were plotted for two values of permissible losses in the appropriate range of Doppler signal shifts. The required speed of the digital components of a two-channel correlator is one-fourth as much than that of a single-channel correlator with identical storage capacities with zero Doppler shift. The advantages in speed and storage capacity when using a two-channel correlator increase significantly as the range of Doppler signal shifts increases.

Thus, two-channel correlation processing circuits have considerably greater efficiency than single-channel circuits when using time quantification of input processes and reference signals.

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GEOPHYSICS, ASTRONOMY AND SPACE

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PROCESSING SIGNALS OF GREAT LENGTH IN ACOUSTIC WAVEGUIDE DEVICES

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 p 70

[Article by A. V. Klimenov, V. D. Markov and V. I. Rogachev]

[Text] A considerable number of papers has now appeared on using digital-coded signals in sonar to detect underwater objects (fish and marine animals) in the presence of a high reverberation noise level. It has been shown that signals in the form of pseudo-random sequences of radio pulses simultaneously provide the best range and velocity resolution. A Sherman sequence may be used as this signal.

The selected signal shaping device may be a pulse sequence driving oscillator which permits Doppler prediction of the emitted signal in a given speed range of the object. It should be expressed in corresponding variation of the time intervals between pulses and the length of the elementary pulse with variation of the carrier frequency. By varying the degree of signal prediction, one can optimally determine the signals from targets moving with specific speed.

The basis of constructing an optimum filter is ultrasonic delay lines in variable cross-section tubular waveguides. These devices provide high value of specific delay time (2-10 ms/m). Constructing the recirculator on the basis of a variable cross-section ultrasonic tubular waveguide provides considerably smaller dimensions of the device. Known recirculators with external feedback tend toward self-excitation and require the use of special stabilizing devices.

To process a signal of 10 pulses lasting 0.5 ms each and with intervals between pulses of 0.5 ms, a recirculator based on a tubular acoustic waveguide with internal coupling by multiple reflection of acoustic waves from the ends of the waveguide was used. These recirculators permit feedback coefficients up to 0.93-0.95.

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GEOPHYSICS, ASTRONOMY AND SPACE

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THE NOISE STABILITY OF AN ANTENNA CONSISTING OF PRESSURE AND FLUCTUATING VELOCITY DETECTORS

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 71-76

[Article by S. N. Zelenskiy and S. V. Pasechnyy]

[Text] It is known that the joint law of the probability distribution of the acoustic velocity values at n points of space may be calculated in approximation of linear acoustics if the joint law of distribution of pressure values at (m + 1)n points of an m-dimensional space (m = 1, 2, 3) is known. This fact leads to the appearance of specific characteristics of the processing system during joint use of fluctuating velocity and pressure detectors. To investigate them, let us find the joint characteristic functional (KhF) of pressure P, velocity $\vec{v}$ and acoustic field sources $\vec{F}$ which by definition is a continual integral of form:

$$\Phi[\varkappa,\vec{f},\vec{\nu}] \overset{\text{def}}{=} \int e^{i[(\varkappa,p)+(\vec{f},\vec{F})+(\vec{\nu},\vec{v})]} dM[p,\vec{F},\vec{v}], \qquad (1)$$

where $\varkappa$, $\vec{f} = (f_1, f_2, f_3)$ and $\vec{\nu} = (\nu_1, \nu_2, \nu_3)$ are arguments of the functional, which are finite functions, $(\varkappa, p) = \int_B \varkappa(R, t)p(R, t)d^3Rdt$ is the scalar derivative, B is a four-dimensional integration domain at which $B = \bar{D} \times T$, $T = (-\infty, \infty)$; $\bar{D}$ is closure of the open domain D of a three-dimensional space with piecewise-smooth boundary $S = \bar{D} \setminus D$; and

$$t \in T,\ R=(R_1,R_2,R_3) \in \mathcal{D},\ (\vec{f},\vec{F}) = \sum_{i=1}^{3}(f_i,F_i);\ (\vec{\nu},\vec{v}) = \sum_{i=1}^{3}(\nu_i,v_i);\ dM$$

is the probability measure of random fields p, $\vec{F}$ and $\vec{v}$.

Let us find the structure of Φ on the basis of the characteristic functional $\mathcal{P}[\varkappa,\vec{f}] \overset{\text{def}}{=} \Phi[\varkappa,\vec{f},0]$ of pressure and source fields $\vec{F}$. Let us consider acoustic fields in media with small viscosity coefficients, described by linear equations of type:

$$\vec{\vartheta}(R,t) \overset{\text{def}}{=} L_1 p + L_2 \vec{F} = \rho_0^{-1}(R) \int_{-\infty}^{t} [\nabla p(R,\tau) + \vec{F}(R,\tau)]\, d\tau, \tag{2}$$

where $\rho_0(R)$ is the density of the medium independent of time. It is further assumed that $\vec{F}$ is equal to zero outside domain B. To determine the functional dependence between ϕ and $\mathcal{T}$, let us determine the properties of operators L_1^* and L_2^*, adjoint to L_1 and L_2. Lemma: Operators L_1^* and L_2^*, which satisfy the relations $(\vec{\vartheta}, L_1 p) = (L_1^* \vec{\vartheta}, p)$ and $(\vec{\vartheta}, L_2\vec{F}) = (L_2^* \vec{\vartheta}, \vec{F})$, have the form:

$$L_1^* = \int_t^{\infty} d\tau [\nabla_R - \vec{\delta}_S] \rho_0^{-1}(R), \quad L_2^* = \rho_0^{-1} \int_t^{\infty} d\tau, \tag{3}$$

where $\vec{\delta}_S$ is the generalized vector function of the simple layer type on surface S with unit density:

$$(\vec{\delta}_S \vec{a}, b) \overset{\text{def}}{=} (\vec{\delta}_S, \int_{-\infty}^{\infty} dt\, \vec{a} \cdot b)_0 =$$
$$= (1, \int_{-\infty}^{\infty} \vec{a} \cdot b \cdot dt)_S = (\vec{a}, b)_{S+T}, \quad (\vec{a}, b)_D \overset{\text{def}}{=} \int_D \vec{a} b\, d^3R, \quad (\vec{a}, b)_{S+T} \overset{\text{def}}{=} \int_{-\infty}^{\infty} dt \oint_S b \cdot \vec{a} \cdot d\vec{S};$$

a and b are the integrated vector and scalar functions, respectively; and $d\vec{S}$ is a vector with direction outside the normal to the surface S.

By integrating in (1) with regard to (2) and (3), we find the solution of the problem in the form:

$$\phi[\varkappa, \vec{f}, \vec{v}] = \mathcal{P}[\varkappa + \int_t^{\infty} d\tau\, \nabla(\rho_0^{-1} \vec{v}), \vec{f} - \rho_0^{-1} \int_t^{\infty} \vec{v}\, d\tau]. \tag{4}$$

Thus, synthesis of optimum processing systems which utilize pressure and fluctuating speed detectors located outside domain D may be carried out by using the derived formula (4) with known field statistics P and forces $\vec{F}$. A special form of expression (4) in the case of constant density of the medium and $\vec{F} = 0$ is found in [1] by continual integration of equations in functional derivatives with respect to the characteristic functional. The method of this paper is based on direct calculation of the mean value using the proved lemma and permits one to find the joint KhF by a less cumbersome method.

Let us consider the structure of the KhF for the case when the acoustic field is subordinate to the wave equation: $\Delta p(R,t) - c^{-2}(R) \frac{\partial^2 p(R,t)}{\partial t^2}$, $R \in D$. Making use of the theorem on divergence and equation (2), at $\vec{F} = 0$, we find the integral equation with respect to p(R, t) of type:

$$p(r,\tau)=\oint_S\int_T\Big[p(R_s,t)\frac{\partial G(R_s,t\,|\,r,\tau)}{\partial n}+\rho_0 G(R_s,t;r,\tau)\frac{\partial v_n(R_s,t)}{\partial t}\Big]dS\,dt, \tag{5}$$

where v_n is the velcocity component normal to the surface S at point $R_s \in S$; and G is solution of the wave equation with addition of the delta-function $\delta(R-r)\,\delta(t-\tau)$ to its right side. Consequently, with regard to (5), formula (4) may be transformed to the following form:

$$\begin{gathered}\Phi[\varkappa,\vec{f},\vec{v}]=\mathcal{P}[\tilde{\varkappa},\vec{f}]=\Phi_s\Big[\Big(\frac{\partial G(R_s,t;r,\tau)}{\partial n_s},\tilde{\varkappa}(r,\tau)\Big),-\rho_0\Big(\frac{\partial G(R_s,t;r,\tau)}{\partial t},\tilde{\varkappa}(r,\tau)\Big)\Big],\\ \Phi_s[\varkappa,\nu]=\langle\exp\{i[(\varkappa,p)_{S\times T}+(\nu,v_n)_{S\times T}]\}\rangle,\\ \tilde{\varkappa}(r,\tau)=\varkappa(r,\tau)+\rho_0^{-1}\int_{\tau}dt\,\nabla\vec{v}(r,t),\quad T=(-\infty,\infty),\quad(\,)_{S\times T}=\oint_S\int_T dS\,dt.\end{gathered} \tag{6}$$

Thus, the field KhF described by a homogeneous wave equation upon observation in volume D is expressed by the joint KhF of pressure p on surface S and the normal velocity component v_n to this surface.

The derived expressions for KhF permit one to determine the ratio of similarity and consequently the structure of optimum detectors when processing signals in both Gaussian and non-Gaussian noise field.

Let us analyze the efficiency of the receiving system for detecting the signal field $s(y,\tau)$, consisting of pressure and velocity detectors for a Gaussian noise field model of acoustic origin. Let us use the spatial detection parameter $d=\langle U_{vykh}\rangle\sigma^{-1}$, i.e., normalized, to the mean square deviation of noise σ, increment of the mean value of the effect at the optimum detector output in the presence of the signal. With regard to (6):

$$d^2=\oint_S dS\int_T d\tau\sum_{k=0}^{1} b_k(y,\tau)\,s_k(y,\tau), \tag{7}$$

where dS is the differential surface S, $s_0(y,\tau)$ and $s_1(y,\tau)$ are the pressure field and normal velocity component to the surface S of a signal at point $y\in S$, respectively; and $b_k(y,\tau)$ and $(k=0.1)$ is solution of a system of integral equations:

$$\oint dS\int_T d\tau\sum_{k=0}^{1}K_{\ell k}(x-y,t-\tau)\,b_k(y,\tau)=s_\ell(x,t),\qquad y,x\in S,\ \ell=0,1, \tag{8}$$

where $K_{\ell k}(x-y,t-\tau)=\langle[U_\ell(x,t)-S_\ell(x,t)][U_k(y,\tau)-S_k(y,\tau)]\rangle$ is a correlation function of the l-th and k-th component and u_0 and u_1 are the pressure field and normal velocity component to surface S, respectively.

Let the signal field be a plane wave:

$$s_k(y,\tau) = A_k(y,\tau)\cos(\omega_0\tau - \vec{\kappa}y), \quad y \in S, \tag{9}$$

and let the correlation function of noise:

$$K_{\ell k}(x-y, t-\tau) = r(t-\tau)\{g_{c\ell k}(x-y)\cos[\omega_0(t-\tau)] + g_{s\ell k}(x-y)\sin[\omega_0(t-\tau)]\} \tag{10}$$

also correspond to the correlation function of the quasi-harmonic equally tinted acoustic field.

System (8) can be solved with regard to (9) and (10) by the method of envelopes [2]. In this case the optimum processing algorithm has approximately the form:

$$b_k(y,\tau) = 2[b_k^{(1)}(y,\tau)\cos(\omega_0\tau) + b_k^{(2)}(y,\tau)\sin(\omega_0\tau)], \tag{11}$$

where $b_k^{(1)} = \mathrm{Re}\tilde{b}_k$, $b_k^{(2)} = \mathrm{I_m}\tilde{b}_k$ and the complex envelope $\tilde{b}_k$ is solution of the integral equation:

$$\begin{gathered}\oint_S dS \int_T d\tau \sum_{k=0}^{3} r(t-\tau)\tilde{b}_k(y,\tau)\tilde{g}_{\ell k}(x-y) = \tilde{\lambda}_\ell(x,0),\\ \tilde{g}_{\ell k}(x) = g_{c\ell k}(x) + i g_{s\ell k}(x), \; \tilde{\lambda}_\ell(x,t) = A_\ell(x,t)e^{-i(\omega_0 t - \vec{\kappa}x)}.\end{gathered} \tag{12}$$

Equation (12) is solved by separation of variables. Having assumed that $b_k(y, \tau) = b_t(\tau)b_{ks}(y)$, from (12) we find the integral equations with respect to space b_{ks} and time b_t optimum processing:

$$\begin{gathered}\oint_S dS \sum_{k=0}^{1} b_{ks}(y)\tilde{g}_{\ell k}(x-y) = e^{i\vec{\kappa}x}[\delta_{0\ell} + (1-\delta_{0\ell})(\rho_0 c_0)^{-1}],\\ \int_T d\tau\, r(t-\tau)b_T(\tau) = A_0, \qquad A_0 = \mathrm{const},\; \delta_{0\ell} = \begin{cases}1, \ell=0\\ 0, \ell=1.\end{cases}\end{gathered} \tag{13}$$

Thus, time processing b_t with joint use of pressure and velocity detectors differs in no way from time processing for a receiving antenna compiled only from pressure detectors. Only the spatial parts of the processing channel are distinguished. Having solved (13) and having substituted (11) into (7), we find:

$$d^2 = d_S d_T,$$

where

$$d_s = \oint_S dS \sum_{k=0}^{1} b_{ks}(y) \frac{\tilde{s}_k^*(y,0)}{A_0}, \quad d_T = \int_T d\tau\, A_0 b_T(\tau), \tag{14}$$

(the symbol * is used to denote complex conjugation operations). We note that the value of d_s is real, despite the complexity of b_{ks} and $\tilde{s}_k$, $k = 0.1$.

Let us compare the noise resistance of an antenna consisting only of pressure detectors with combined antenna consisting of long-range fluctuating velocity detectors based on analysis of the spatial parameter d_s. Let surface S be a rectangular round cylinder of radius ρ and length 1. Let us assume that $\rho \ll 1$; $\rho \ll R_{kor}$, where λ is the signal wavelength and R_{kor} is the noise field correlation radius, and the fluctuating velocity channels also contain additive white Gaussian noise along with acoustic noise. It follows from (13) in this case that the first term of expansion by powers of ρ of optimum space-time processing corresponds to receiving system in the form of an optimum processing channel, which is an antenna with length 1 consisting of pressure detectors and fluctuating velocity component detectors orthogonal with each other and with respect to the cylinder generatrix. In the indicate, sense, the optimum detector based on a linear base of point, combined pressure and velocity detectors is equivalent to an optimum detector with the same detectors located at S.

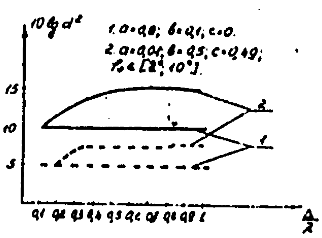

Figure 1

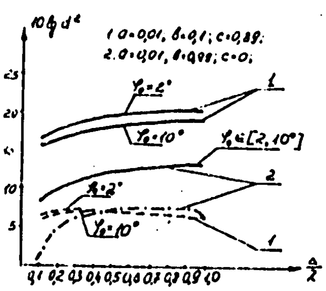

Figure 2

Graphs of the dependence of the space detection parameter d_s on the ratio Δ/λ of a linear equidistant antenna consisting of n = 10 and combined detectors, where Δ is the distance between adjacent detectors, are plotted by the solid lines in Figures 1 and 2. The noise field is the sum of independent isotropic and anisotropic acoustic noise and also noise δ-correlated in space [3] and [4]:

$$\tilde{g}_{00}(\rho_{ij}) = a\delta_{ij} + b\,\frac{\sin\left(\frac{\omega_0}{c_0}\right)\rho_{ij}}{\frac{\omega_0}{c_0}\rho_{ij}} + c\,\frac{J_1\left(\frac{\omega_0}{c_0}\rho_{ij}\sin\varphi_0\right)}{\frac{1}{2}\frac{\omega_0}{c_0}\rho_{ij}\sin\varphi_0},$$

$$\tilde{g}_{01}(\rho_{ij}) = 0,$$

$$\tilde{g}_{11}(\rho_{ij}) = (\rho_0 c_0)^{-2}\left[a\delta_{ij} + b\,\frac{\sin\left(\frac{\omega_0}{c_0}\rho_{ij}\right) - \frac{\omega_0}{c_0}\rho_{ij}\cos\left(\frac{\omega_0}{c_0}\rho_{ij}\right)}{\left(\frac{\omega_0}{c_0}\rho_{ij}\right)^2} + 2c\,\frac{J_2\left(\frac{\omega_0}{c_0}\rho_{ij}\sin\varphi_0\right)}{\left(\frac{\omega_0\rho_{ij}}{c_0}\right)^2}\right],$$

where i, j = 1, 2, ... n; a, b, c = const; c_0 is the speed of sound; $\rho_{ij} = (i-j)\Delta$ is an antenna oriented along the y-axis; the wave vector of the signal is parallel to the x-axis; the anisotropic field arrives from directions whose latitude φ is less than φ_0 and the spectral density of the pressure field output depends on φ by the law: $\cos\varphi$; $\varphi_0 \leq \pi/2$; and I_1 and I_2 are cylindrical functions of first kind and first and second order, respectively.

The graphs of $d\Omega(\Delta/\lambda)$ for an antenna consisting of pressure detectors are plotted by the dashed lines in Figures 1 and 2 for comparison. We note that the advantage of an antenna consisting of pressure and velocity detectors may reach approximately 14 dB with significant prevalence of an anisotropic component (c = 0.99).

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THE NOISE STABILITY OF SIMPLE AND OPTIMUM CROSS-CORRELATION DETECTORS

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 78-86

[Article by Ye. A. Danilova, N. A. Drozdova and V. V. Ol'shevskiy]

[Text] The classical theory of optimum signal detection systems on a noisy background now developed (see, for example, [1-4]) permits strict determination of the processing algorithm of input processes if the complete probability description is given for them. Processing algorithms and the structures of detectors were determined by using optimum detection theory with respect to hydroacoustic data [5-13] with the following typical model of the input process

$$X(t)=S(t)+F(t)+N(t),$$

where the echo-signal S(t) is a pulse of precisely known shape with random initial phase φ uniformly distributed in the range (0.2π);

$$S(t)=\dot{A}C_0(t)\cos[(\omega_0+\Omega)t+\Phi(t)+\varphi],$$

where $C_0(t)$ is the envelope of the emitted signal; ω_0, $\Phi(t)$ are the emitted signal carrier frequency and a function which determines its phase modulation, respectively; A is a constant value which characterizes the level of the echo-signal; Ω is the value of the Doppler shift of the central echo-signal frequency*; noise is a Gaussian random process consisting of an additive mix of white noise N(t) and marine reverberation F(t).

The structure of an optimum detector corresponding to the model of the input process mentioned above is presented in Figure 1, from which it is obvious that it is a quadrature two-channel correlator which uses functions $g_c(t)$ and $g_b(t)$ as the reference signals, which are found as solutions of the following integral equations:

*The simplest variant of the Doppler effect -- a frequency shift of the echo-signal spectrum -- is considered here and further, although in fact the Doppler effect is manifested multiplicatively in the form of a multiplier in front of the time argument when studying a complex signal.

$$\left.\begin{aligned} S_c(t) &= \frac{G_w}{2} g_c(t) + \int_{-T}^{T} B_p(t-u)\, g_c(u)\, du; \\ S_s(t) &= \frac{G_w}{2} g_s(t) + \int_{-T}^{T} B_p(t-u)\, g_s(u)\, du, \end{aligned}\right\} \tag{1}$$

where $S_c(t)$ and $S_s(t)$ are the quadrature components of the echo-signal, $G_w/2$ is the spectral density of the "white" Gaussian noise output, $B_p(\tau)$ is the reverberation correlation function:

$$B_p(\tau) = \frac{I_p}{E_u} \int_{-\infty}^{\infty} C(t)\, C(t-\tau)\, dt, \tag{2}$$

where I_p is the mean reverberation intensity, E_u is the energy of the emitted signal and $C(t)$ is the emitted signal.

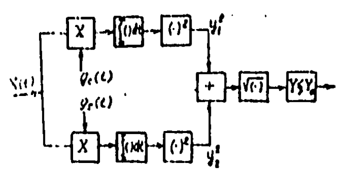

Figure 1. Optimum Detector of Echo-Signals of Known Shape With Random Initial Phase on Background of Mixture of "White" Noise and Reverberation

The value of Y shaped at the detector output has the form:

$$Y = \sqrt{\left(\int_{-\infty}^{\infty} X(t)\, g_c(t)\, dt\right)^2 + \left(\int_{-\infty}^{\infty} X(t)\, g_c(t)\, dt\right)^2} \tag{3}$$

Let us introduce the following parameters [8]:

$$Q_{\text{рш}0} = \frac{I_{p0} T_0}{G_w/2} \tag{4}$$

-- a parameter which characterizes the ratio of the mean reverberation energy to the spectral noise output:

$$Q_{\text{эш}0} = \frac{I_{\text{э}0} T_0}{G_w/2} \tag{5}$$

-- a parameter which characterizes the ratio of the mean echo-signal energy to the spectral noise output and in this case $I_{po} = I_p\ (T_{ef} = T_o)$; $I_p = \langle F^2(t) \rangle$:

$$\alpha = \Delta F_{\text{эф}} T_0 \tag{6}$$

-- the relative effective width of the signal spectrum;

$$\beta = T_{ef} / T_0 \tag{7}$$

-- its relative effective length;

$$\nu = \frac{\Omega T_0}{2\pi} \tag{8}$$

-- the relative Doppler shift of the echo-signal frequency.

In this case T_0 = const is some fixed length of the emitted signal at which the initial characteristics of the echo-signal and noise are determined, T_{ef} is the effective length of the emitted signal and ΔF_{ef} is the effective bandwidth of the emitted signal.

Solving equation (1) with regard to relations (2), (4)-(8), it is easy to find the following representation of the quadrature components of the reference signal:

$$\left.\begin{aligned} g_c(t) &= \frac{\sqrt{Q_{sho}}}{2\pi\sqrt{G_{sh}}T_0}\int_{-\infty}^{\infty}\frac{\varphi_{kc}\left(\omega - \frac{2\pi\nu}{T_0}\right)\exp(j\omega t)\,d\omega}{1 + \frac{Q_{rsho}\beta}{T_{ef}T_0}|\varphi_{kc}(\omega)|^2}; \\ g_s(t) &= \frac{\sqrt{Q_{sho}}}{2\pi\sqrt{G_{sh}}T_0}\int_{-\infty}^{\infty}\frac{\varphi_{ks}\left(\omega - \frac{2\pi\nu}{T_0}\right)\exp(j\omega t)\,d\omega}{1 + \frac{Q_{rsho}\beta}{T_{ef}T_0}|\varphi_{ks}(\omega)|^2}, \end{aligned}\right\} \tag{9}$$

where φ_{ks} and φ_{is} are the spectra of the cosine and sine components of the emitted signal, respectively.

As follows from the relations given above, one must know precisely the ratio Q_{rsho} of reverberation interference to noise and also the Doppler parameter ν to realize an optimum detector and one must shape his own reference signals $g_c(t)$ and $g_s(t)$ according to algorithms (9) for each combination of these parameters. It is obvious that this detector is sufficiently complex. Therefore, the problem of how much better an optimum detector is than a so-called simple correlator [6, 10-12], in which the quadrature components of the emitted signal obtained by substitution of $Q_{rsho} = 0$ into expression (9) are used as the reference signals $g_c(t)$ and $g_s(t)$:

$$\left.\begin{aligned} g_c(t) &= C_c(t)/G_{sh}, \\ g_s(t) &= C_s(t)/G_{sh}. \end{aligned}\right\} \tag{10}$$

We note that along with a detector optimum in the strict sense, a so-called Urkowitz filter, which is one of the variants of a "bleaching" filter, is sometimes discussed [1, 3, 5, 10]. It is assumed in this case that a Urkowitz filter, although it is not an optimum detector in the strict sense, is very close to it.

Let us make a brief survey of the known papers [6, 10-12] devoted to comparison of the quality of certain detectors with different models of input processes and types of emitted signals.

The problem of calculating the optimum signals and detectors for an active sonar system operating in the presence of reverberation and "white" noise is considered in [6]. The motion of the location object and the motion of the scatterers with given constant velocity are taken into account in this case. The structures and characteristics of the quality of the optimum and suboptimum detectors are found.

The following value is used as the detection parameter

$$\rho = \frac{m^2(Y/1)}{d(Y/1)}, \tag{11}$$

which characterizes the signal/noise ratio at the detector output and m(Y/1) and d(Y/1) are the mean value and standard deviation of Y in the presence of an echo-signal, respectively.

The following results were found on the basis of comparing the characteristics of the quality of optimum and suboptimum systems:

-- at zero speeds of the location object, an increase of the bandwidth of the emitted signal yields uniform improvement of the characteristics of quality.

-- with non-zero speeds of the location object, it is recommended that either simple signals or signals with very great bandwidth be used since characteristics of quality are deteriorated due to the effect of reverberation at mean bandwidths.

-- the difference between optimum and suboptimum systems increases monotonically with an increase of the reverberation level. According to the author, the average distance between the systems comprises 1.5-2 dB in this case.

The structure is determined and the noise stability of an optimum detector of signals on a background of an additive mixture of noise and reverberation interference is determined in [10]. In this case the Doppler effect in the echo-signal is not taken into account. The noise stability of an optimum detector with an Urkowitz filter and simple correlator is compared.

The value of ρ, determined according to (11), is taken as the detection parameter. It was found that an Urkowitz filter is inferior in noise stability to an optimum detector which has an input process at a final time interval, with ratio of mean reverberation energy to spectral noise density $Q_{rsh} < 5.56$ dB. At values of $Q_{rsh} > 5.56$ dB, a Urkowitz filter, according to the results of [10], has better noise stability compared to an optimum detector.

It was found for a simple correlator that even with significant excess of reverberation interference over noise, it is slightly inferior in noise stability to an optimum detector (by approximately 1.25 dB).

The problem of deterioration of the quality of signal detection by a simple correlator operating in the presence of reverberation is considered in [11]; In this case the Doppler effect is not considered in the echo-signal.

The value of ρ mentioned above is the detection parameter. The dependence of parameter ρ on the ratio of reverberation energy to the spectral noise density Q_{rsh} is investigated. It is shown in this paper that the use of a simple correlator in the presence of reverberation leads to a 2.17-dB loss in the equivalence of energy and spectral density of noise output Q_{rsh}. It is shown in this paper that the use of a simple correlator in the presence of reverberation leads to a 2.17-dB loss in the equivalence of energy and spectral density of noise output ($Q_{rsh} = 0$ dB) and to an 8.78-dB loss at $Q_{rsh} = 10$ dB.

An optimum signal detection algorithm on a "white" noise and reverberation background is considered in [12] as one of the possible ways of improving detection quality. It is found that the advantage in threshold signals with respect to a simple correlator does not exceed 2.17 dB at $Q_{rsh} < 0.9$ dB. However, this restriction on the smallness of parameter Q_{rsh} may be removed. One of the methods of removing this restriction is proposed in the considered paper.

The results of the investigations considered above [6, 10-12], we feel, do not permit correct comparison of the quality of an optimum detector (in the strict sense) and of suboptimum detectors. The impossibility of this comparison includes the following:

First, parameter ρ used in the indicated investigations and which characterizes the quality of signal detection on a noisy background, although it is monotonically related to the probability of detection, does not yield a clear idea of the separate effect of noise and reverberation noise on noise stability.

Second, the models of useful signals and noise differ somewhat from each other in different papers; reverberation is considered in [6] with regard to random motion of scatterers, whereas the indicated motion is not taken into account in [10-12]; the Doppler shift of frequency in the echo-signal is also considered in [6] and the object of location is regarded as fixed in [10-12].

Third, comparison of the noise stability of different detection systems is difficult due to the fact that emitted signals of different types (signals with square-wave envelope in [10-12] and signals with bell-shaped envelope and linear modulation frequency in [6]) are considered in the noted papers.

Fourth, a Urkowitz filter which, according to the calculutions, better reflects an optimum detector in some cases, is discussed as one of the variants of detectors in [10]. We feel that this result is somewhat doubtful from the viewpoint of the equivalence of input data in solution of the problem of comparing the noise stability of different sonar data processing systems.

In this regard let us consider the problem of comparing the characteristic of quality of an optimum and simple correlator in more detail on the basis of the equivalent parameters (4)-(8) of emitted signals, echo-signals, noise and reverberation noise introduced above.

The expression for detection probability P_0, according to [9], has the form

$$P_0 = \frac{1}{2}\left\{1 + \Phi\left[\sqrt{\rho} - \sqrt{2 \ln \frac{1}{P_{\text{лт}}}}\right]\right\}, \tag{12}$$

where $\Phi(x) = \frac{2}{\sqrt{2\pi}} \int_0^x e^{-t^2} dt$ is the probability integral and P_0 and P_{lt} are the probabilities of detection and false alarms, respectively.

Let us consider a signal with bell-shaped envelope and linear frequency modulation as the emitted signal:

$$C(t) = \exp\left(-\frac{\pi t^2}{2T_{\text{эф}}^2}\right)\cos\left[\omega_0 t + \frac{\Delta\Omega_H}{2T_{\text{эф}}} t^2\right]. \tag{13}$$

We note that in principle there are two cases of emission: constant output of the emitted signal and constant energy of the emitted signal. However, one of these cases -- constant emitted output -- is considered in this paper.

With regard to relations (12) and (13), the dependence of threshold signals on the emitted signal parameters for optimum and simple correlators, respectively, assume the following form:

$$Q_{\text{эшо}}^{(0)} = \frac{\sqrt{\pi}\left[\Phi^{-1}(2P_0 - 1) + \sqrt{2 \ln \frac{1}{P_{\text{лт}}}}\right]^2}{2\beta \int_{-\infty}^{\infty} \frac{\exp\left[-\left(\omega - \frac{\nu\sqrt{\pi}}{\alpha}\right)^2\right]}{1 + \frac{\nu_{\text{рев}}\beta}{\alpha}\exp(-\omega^2)} d\omega}, \tag{14}$$

$$Q^{(t)}_{\text{вых}} = \frac{\left[\Phi^{-1}(2P_0-1)+\sqrt{2\ln\frac{1}{P_{lt}}}\right]^2}{2\beta} \times \left[\frac{Q_{rsho}\beta}{\alpha\sqrt{2}}\exp\left(-\frac{\pi\nu^2}{2\alpha^2}\right)+1\right]. \qquad (15)$$

Based on formulas (13) and (14), numerical calculation of threshold signals were made on the BESM-6 and MIR-2 computers as a function of the emitted signal parameters α and β, the ratio of reverberation noise to a noisy signal Q_{rsho} and of relative Doppler shifts in the echo-signal ν. The results of the calculations are presented in Figures 2-7.

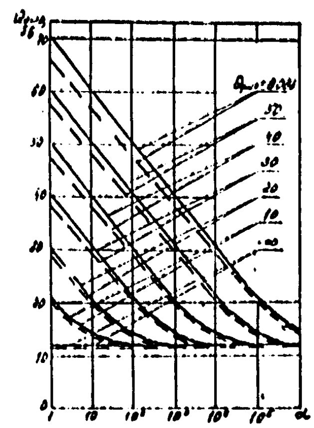

Figure 2. Threshold Signals at $P_0 = 0.9$; $P_{lt} = 10^{-4}$; $\beta = 1$; $\nu = 0$; – – – – – -- optimum detector; ——— -- simple correlator

The given results permit the following conclusions.

The threshold signals decrease for both an optimum system and for a simple correlator in the absence of a Doppler shift in the echo-signal with an increase of parameter α (Figure 2); in this case the difference in the threshold signals for both systems increases as the reverberation level increases but it does not exceed a value of 3 dB for the worst studied case ($\alpha = 1$, $\beta = 1$, $Q_{rsho} = 60$ dB) and for this case the optimum detector and simple correlator are essentially not distinguished in threshold signals if α satisfies the condition $10 \lg \alpha > Q_{rsho}$.

In the presence of a Doppler shift in the echo-signal, the threshold signals have a maximum at specific values of α (see Figures 3-7) both for an optimum detector and for a simple correlator and in this case the position of the maximum threshold signal corresponds by argument α to the condition $\alpha = 3\nu$ (at $\beta = 1$) (the value of the maximum threshold signals for an optimum detector corresponds to a somewhat greater value of parameter α than for a simple correlator).

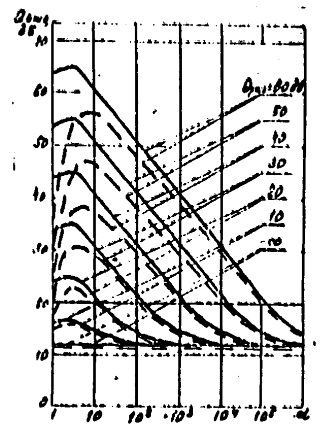

Figure 3. Threshold Signals at $P_0 = 0.9$; $P_{1t} = 10^{-4}$; $\beta = 1$; $\nu = 1$; — — — — -- optimum detector; ——— -- simple correlator

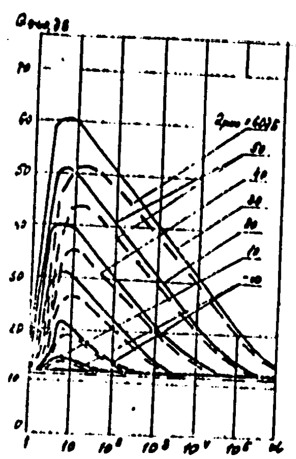

Figure 4. Threshold Signals at $P_0 = 0.9$; $P_{1t} = 10^{-4}$; $\beta = 1$; $\nu = 3$; — — — — -- optimum detector; ——— -- simple correlator

The greatest difference in the threshold signals is observed in the range of the indicated maximum threshold signals and this difference increases with an increase of the reverberation noise level and reaches values of 15-20 dB for the worst studied cases ($\nu \in (1, 100)$, $\beta = 1$ and $Q = 60$ dB.

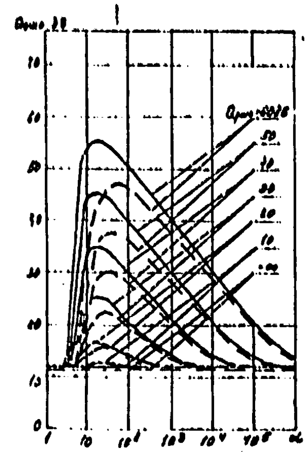

Figure 5. Threshold Signals at $P_0 = 0.9$; $P_{1t} = 10^{-4}$; $\beta = 1$; $\nu = 10$; – – – – -- optimum detector; ——— -- simple correlator

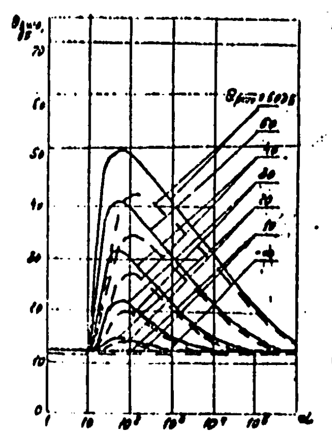

Figure 6. Threshold Signals at $P_0 = 0.9$; $P_{1t} = 10^{-4}$; $\beta = 1$; $\nu = 30$; – – – – -- optimum detector; ——— -- simple correlator

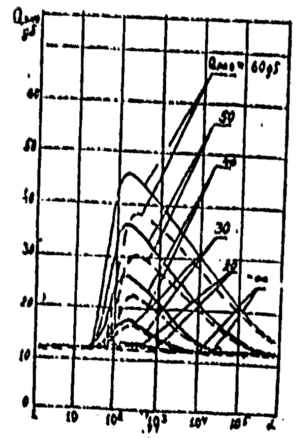

Figure 7. Threshold Signals at $P_0 = 0.9$; $P_{1t} = 10^{-4}$; $\beta = 1$; $\nu = 100$; ------ optimum detector; ——— -- simple correlator

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GEOPHYSICS, ASTRONOMY AND SPACE

UDC 612.391.16

THE PROBLEM OF PROCESSING A SIGNAL FIELD IN A RANDOMLY INHOMOGENEOUS MEDIUM

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 87-93

[Article by S. N. Zelenskiy and S. V. Pasechnyy]

[Text] The properties of the probability density of an acoustic field and the structure of the similarity ratio in acoustic signal detection in a randomly inhomogeneous medium are investigated in the given paper. Let us use the expression for the characteristic functional (KhF) of an acoustic pressure field which is formed by a statistically independent signal and noise in the form [1]:

$$P[æ]=\exp\{i(p_0,æ)\}\,\theta_n[æ]\,\theta_c[æ], \qquad (1)$$

where $i^2=-1$; $\theta_c[æ]=\Phi_{s\sim f}[\vec{\sigma}[æ];0,0]$ is the KhF of signal fluctuations, $\Phi_{s\sim f}$ is the KhF of random inhomogeneities, $\theta_p[\mathscr{H}]$ is the KhF of the noise field, p_0 is the mean value of the signal field in the absence of random inhomogeneities, the parentheses (,) denote the scalar derivative,

and (R, t) is the argument of the KhF. Let us further imply that $P[\mathscr{H}]$ given by expression (1) has meaning either as a functional of $\mathscr{H}(.,.)$ or as the characteristic function of an n-dimensional vector:

$$\vec{æ}=(æ(R_{(1)},t_{(1)})\delta(R_{(1)}-R)\delta(t_{(1)}-t),\ldots,æ(R_{(N)},t_{(N)})\delta(R_{(N)}-R)\delta(t_{(N)}-t)),\ N=1,2,\ldots$$

and, consequently, its Fourier transform has meaning of probability density, R and $R_{(k)}$ are three-dimensional radius vectors of spatial points in Cartesian coordinates and t and $t_{(k)}$ are moments of time k = 1, 2, 3 ...

Let us find the expression for the N-dimensional probability density $\omega(\vec{\nu})$ of the acoustic field, for which we expand the KhF $\theta_s[æ]$ to a functional series with powers of $\mathscr{H}$ and let us derive the inverser Fourier transform of KhF $P(æ)$:

$$w(\vec{\nu}) = \theta_c\left[i\frac{\delta}{\delta\vec{\nu}}\right] w_{on}(\vec{\nu}), \quad \vec{\nu} = (\nu_1, \ldots, \nu_N), \tag{2}$$

where

$$\frac{\delta}{\delta\vec{\nu}} = \left(\delta(\nu - \nu_1)\frac{\partial}{\partial\nu_1}, \ldots, \delta(\nu - \nu_N)\frac{\partial}{\partial\nu_N}\right), \quad \nu = \nu(R, t),$$

$w_{op}(\vec{\nu})$ is the Fourier transform of the functional $\exp\{i(p_0, \varkappa)\}\,\theta_p[\varkappa]$.

Expression (2) is an expansion of density $\omega(\vec{\nu})$ into a series by statistical moments of signal field fluctuations. To determine the functional dependence of $w(\vec{\nu})$ on $\vec{\nu}$ (by expansion of $w(\vec{\nu})$ into a Taylor series), one can use the following variety of (2):

$$w(\vec{\nu}) = w_{on}\left(\vec{\nu} + i\frac{\delta}{\delta\vec{\varkappa}}\right)\theta_c[\varkappa = 0]. \tag{3}$$

At the same time formula (2) may be preferable when determining the effect of the statistical moments of the signal field on the form of w.

Let us find yet another pair of formulas for $w(\vec{\nu})$ for which we represent the KhF $\theta_p[\varkappa]$ in the form of a functional Taylor series. After transformation, it can be written in the form:

$$w(\vec{\nu}) = \theta_n\left[i\frac{\delta}{\delta\vec{\nu}}\right] w_{oc}(\vec{\nu}), \tag{4}$$

$$w(\vec{\nu}) = w_{oc}\left(\vec{\nu} + i\frac{\delta}{\delta\vec{\varkappa}}\right)\theta_n[\varkappa = 0], \tag{5}$$

where $w_{os}(\vec{\nu})$ is an N-dimensional Fourier transform of the functional $\exp\{i(p_0, \varkappa)\}\,\theta_s[\varkappa]$. The same as (2), expression (4) is an expansion of multidimensional probability density into a series, but in the given case by the statistical moments of the noise field. Similar to formula (3), expression (5) should be used when determining the functional dependence of w on $\vec{\nu}$.

We note that expressions (2) and (3) must be used in known noise field statistics w_{op} and with not completely known w_s, whereas expressions (4) and (5) must be used in the opposite case -- the known statistics of the signal field w_{os} and not completely known statistics of the noise field w_p.

Let us generalize formulas (2)-(5) to the case when the signal field p_s and noise field p_p are statistically related, for which we represent p_p in the form of two terms of statistically independent p_p and dependent p_{pg} with the signal field, i.e., $p_p = p + p_{pg}$. In this case, for the case of (2), we find:

$$w(\vec{\nu}) = \theta_c\left[i\frac{\delta}{\delta\vec{\nu}} / i\frac{\delta}{\delta\vec{\mu}}\right] w_{on}\left(\vec{\nu} / i\frac{\delta}{\delta\vec{\mu}}\right)\theta_{ny}(\mu)\Big]_{\mu=0}, \tag{6}$$

where $\theta_s[\varkappa/p_{pz}]$ and $w_{op}(\vec{\nu}/p_{pz})$ are the arbitrary characteristic functionals of the signal field p_s and the probability density of component p_p of the noise field, respectively (with fixed value of statistically dependent component p_{pg}). Expression (6) is an expansion of probability density by the field correlation moments p_{pz}. Component p_{pz} is usually small and may operate in (6) only by several terms for the coupling moments. Formulas (3)-(5) may be generalized in like fashion. Let us use formulas (2)-(5) to determine the plausibility ratio and to determine the structure of an optimum space-time processing channel. Let the noise field be Gaussian. In this case it is more convenient to use expression (3), which permits representation of the ratio of plausibility $\Lambda_{(k)}$ in the form:

$$\Lambda_{(N)} = \Lambda_{o(N)} \exp\left\{-\frac{1}{D}\sum_{j=1}^{N}\sum_{k=1}^{N} D_{jk}\sigma_j^{-1} i\frac{\partial}{\partial\varkappa_j}\frac{\nu_k + \frac{1}{2} i\frac{\partial}{\partial\varkappa_k} - p_{ok}}{\sigma_k}\right\}\theta_c[\vec{\varkappa}=0], \tag{7}$$

where $\Lambda_{o(N)}$ is the ratio of plausibility upon detection of the determined signal component p_o on a Gaussian noise background with KhF $\theta_{nz}[\varkappa]$;

; $\varkappa_k$ is the k-th component of vector $\vec{\varkappa}$; matrix

$D_{jk}[D\sigma, \sigma_k]^{-1}$ is the inverse to the correlation matrix of the noise field;

and $p_{ok} \overset{Df}{=} p_o(R_{(k)}, t_{(k)})$.

Expression (7) may be transformed to the following form:

$$\Lambda_{(N)} = \Lambda_{o(N)}\theta_c\left[i\frac{\delta}{\delta\vec{\mu}}\right]\exp\left\{-\frac{1}{D}\sum_{k=1}^{N}\sum_{j=1}^{N} D_{jk}\sigma_j^{-1}\mu_j\frac{\nu_k + \frac{1}{2}\mu_k - p_{ok}}{\sigma_k}\right\}, \tag{8}$$
$$\text{at } \vec{\mu}=0$$

where $\vec{\mu} = (\mu_1, \ldots \mu_n)$.

Unlike (7), formula (8) is explicit representation of the plausibility ratio in the form of a series by statistical moments of signal field fluctuations. In the case of Gaussian KhF θ_s, formula (7) or (8) is a series by powers of $B_s B_p^{-1}(\nu - p_o)B_p^{-1}(\nu - p_o)$, where B_s and B_p are correlation matrices of the signal and noise fields, respectively.

In the regular case in the range of $N \to \infty$, from (7) we find the following expression for the functional of the plausibility ratio Λ:

$$\Lambda = \Lambda_0 \exp\{-(\nu + \frac{1}{2} i \frac{\delta}{\delta\varkappa} - p_0, B_n^{-1} i \frac{\delta}{\delta\varkappa})\} \theta_c[\varkappa = 0], \tag{9}$$

where B_p^{-1} is an operator inverse to the correlation operator of the noise field B_p. Let us introduce the notation $D = \delta/\delta\varkappa$; then operator $\exp(\nu, D)$ is the operator of the shift by value ν on a set of analytical functionals of argument $\varkappa$. Having denoted the order of action of D and $\varkappa$ in formula (9), according to [2], after transformations we find:

$$\Lambda = \Lambda_0 \theta_c[\overset{1}{\varkappa} + \overset{2}{\mathcal{D}} B_n^{-1} - i(\nu - p_0) B_n^{-1}]\Big|_{\varkappa=0}$$
$$\overset{1}{\varkappa}^k \cdot \overset{2}{\mathcal{D}}^l \cdot \overset{3}{\mathcal{D}}^m \equiv \mathcal{D}^m \mathcal{D}^l \varkappa^k; \quad \mathcal{D} = \frac{\delta}{\delta\varkappa}; \quad k, l, m = 0, 1, 2, \ldots \tag{10}$$

The output effect of an optimum detector, understood as $l_p \Lambda$, according to (10), is equal to the sum of the output effects of an optimum detector for detecting a determined signal component p_0 and an optimum detector for detecting a stochastic signal component to whose input the realization $\nu - p_0$ arrives. The optimum processing channel is shown schematically in Figure 1.

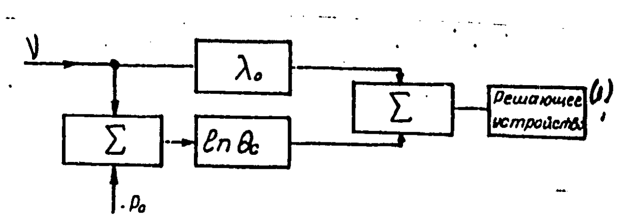

Figure 1

KEY: 1. Resolving device

Let us consider in more detail the structure of a channel for processing the fluctuating component of the signal which is formed only due to passage of a determined wave packet p_0 through a random small-scale weakly inhomogeneous medium. Let

$$p_0(R,t) = p_m(t, \frac{R}{c}) \cos(\omega t - \kappa R),$$
$$p_m = \begin{cases} 1, & t_1(R) \leq t \leq t_2(R), \\ 0, & t \in [t_1(R), t_2(R)], \end{cases} \tag{11}$$

$[t_2(R) - t_1(R)]$ is the pulse length at point R, R = (x, y, z). Also let the Born approximation -- approximation of single scattering -- on the entire range of propagation -- be unacceptable; at the same time it may be used at shorter distances during passage of which the initially determined wave packet becomes random with asymptotically normal probability distribution law. We will call wave packet (11) conditionally Gaussian after passage through this domain.

Let us dwell on the problem of the possibility of Gaussian approximation of the signal field for a medium with independent discrete random inhomogeneities of the speed of sound. Let:

$$c^{-2} = c_0^{-2}(1+\varepsilon); \; c_0^{-2} = \langle c^{-2} \rangle,$$

where ε is a random function with uniform distribution law in the range $(-b, b)$, $b > 0$. Using the results given in [6], one can show that the error obtained as a result of replacing the true integral probability distribution law F by Gaussian law F_g can be estimated by the formula

$$|F_r[p] - F[p]| < \frac{3^{3/2}}{4} C_r \frac{\bar{d}^3}{\tilde{d}^3} \cdot \frac{V}{\ell^2 \Delta}, \tag{12}$$

where

$$\bar{d}^3 \simeq n^{-1}\sum_{k=1}^{n} d_k^3, \; \tilde{d}^2 \simeq n^{-1}\sum_{k=1}^{n} d_k^2, \; n \approx \frac{\Delta \ell^2}{V},$$

$d_k^3 = \langle |p_k|^3 \rangle \langle |p_k|^3 \rangle^{-1}$, p_k is a field scattered by the k-th inhomogeneity, $(2\pi)^{-1/2} \leq c_g \leq 0.905$, V is the average volume occupied by the inhomogeneity, $l^3 V^{-1}$ is the number of scatterers from which (provided they are identical) the scattered fields with approximately equal amplitude and phase arrive at the reception point and Δ is the thickness of the wave packet. In the case of a weakly inhomogeneous continuous medium, one must substitute ρ^3, where ρ is the correlation radius, instead of V to estimate the error of Gaussian approximation. With an increase of the thickness of the wave packet Δ, as follows from (12), the approximation error is reduced, approaching zero at $\Delta \to \infty$. However, at large values of Δ, formula (12) is unsuitable since the condition of applicability of Born approximation [3], to which may be given the form: $\Delta \ll 3\omega^{-2} b^{-2} V^{-1/3} c^{-2}$, is violated. Consequently, after the leading edge of the wave packet covers distance $x \overset{Df}{=} kR \geqslant x_0 = 0.3\omega^{-2} b^{-2} V^{-1/3} c^2$, the signal probability distribution law becomes arbitrarily Gaussian. It is obvious that the type of probability distribution law of the acoustic field values on surface x = const depends on the value of x, and, consequently, the structure of the optimum channel which realizes θ_s depends on the values of x which determine the spatial arrangement of the receiving antenna. Let the antenna be located inside the domain of space $x_1 \leqslant x \leqslant x_2$, whose dimensions in the direction of the signal propagation vector k are much less than x_0: $x_2 - x_1 \ll x_0$. Then

$lp\,\theta_s$ is function x and functional ν. At $k^{-1} \lesssim x \lesssim x_0$, $l_p\,\theta_s$ is realized by means of the well-known channel [4] containing the correlator.

Let us consider the case of $x_0 \lesssim x \lesssim 2x_0$, which corresponds to passage of a signal with Gaussian distribution law through a randomly inhomogeneous medium. In this case:

$$p(R,t) = p_0(R,t) + S_2(R,t)(p_0 + p_1) + p_1(R,t),$$

p_1 is a field scattered by layer $\lambda \lesssim x \lesssim x_0$ and has normal probability distribution law, $p_2 \overset{\mathrm{Df}}{=} S_2(p_0 + p_1)$ is a field scattered by the second layer with fixed realization p_1 and has normal probability distribution law, $S_2'(R, t)p \overset{\mathrm{Df}}{=} \int d^3r d\tau\, g(R, t; r, \tau)p(r, \tau)$, $g(\cdot)$ is the product of the Green function by the value of the fluctuations of random inhomogeneities and $x_0 \lesssim kR \lesssim 2x_0$. Within the considered approximation:

$$\exp\left\{-\frac{1}{2}(A_2 \varkappa ; \varkappa)\right\} \simeq 1 - \frac{1}{2}(A_2 \varkappa, \varkappa),$$

where $A_2 = \langle S_2' S_2'' \rangle \nu' \nu''$ and the primes indicate on which function S_2 acts: $A_2 \equiv \langle (S_2 \nu) - (S_2 \nu) \rangle$. After transformation of expression (1), using the Novikov-Furtz formula [5], we find:

$$\begin{aligned} \theta_c[\varkappa] = {} & \exp\left\{-\frac{1}{2}(B_2 \varkappa, \varkappa)\right\} \exp\left\{-\frac{1}{2}(B_1 \tilde{\varkappa}, \tilde{\varkappa})\right\} \times \\ & \times \Big\{1 - \frac{1}{2}(\langle g(I,1) g(II,2) \rangle \varkappa(I) \varkappa(II) [B_1(2,1) - \\ & - (\tilde{\varkappa}(3) \tilde{\varkappa}(4), B_1(3;1) B_1(4;2))], 1)\Big\}, \end{aligned} \tag{13}$$

where $B_1 = \langle p_1(R,t) p_1(r,\tau) \rangle$, $B_2 = \langle S_2'(R,t) S_2''(r,\tau) \rangle p_0' p_1''$, and $(\cdot, 1)$ is the notation of scalar multiplication by unity in the functional space; $\tilde{\varkappa}(R,t) = \varkappa(R,t) + i \int d^3R^{(1)} dt^{(1)} d^3R^{(2)} dt^{(2)} \varkappa(R^{(1)}, t^{(1)}) \times \varkappa(R^{(1)}, t^{(1)}) \langle [S_2(R^{(1)}, t^{(1)}) p_0] g(R^{(2)}, t^{(2)}; R, t) \rangle$ and the identical numbers serve to denote identical arguments of the functions. It follows from (13) that the channel which processes the signal fluctuation component is significantly nonlinear, i.e., it is not reduced to the linear circuit and correlator.

Let us expand (13) by powers of the product of the exponents and let us analyze only the first two terms of the series. This corresponds to slight deviation of the distribution law from Gaussian. In this case the optimum processing channel $lp\,\theta_s$ is broken down into two channels: a Gaussian signal processing channel with correlation operator $B_1 + B_2$ and a channel which takes into account the non-Gaussian nature of the signal fluctuation distribution law, containing second- and fourth-order nonlinear circuits by variable $\nu - p_0$. In the general case, the separation mentioned above does not occur in explicit form.

Thus, the developed method of calculating the ratio of plausibility permits one to investigate the properties of optimum signal field detectors in a randomly inhomogeneous medium with deviation of the probability distribution law from Gaussian.

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GEOPHYSICS, ASTRONOMY AND SPACE

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EFFECT OF SEA REVERBERATION ON SIGNAL DISCRIMINATION CHARACTERISTICS

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 98-99

[Article by A. K. Senatorov and A. P. Trifonov]

[Text] The optimum (in general form) solution of the problem of discriminating signals scattered in the sea medium causes considerable difficulties related to the absence of a priori information about the parameters of the sea medium which characterize the reverberation process $V(t)$ and noise of the sea $n(t)$. In view of this, let us investigate the effect of sea reverberation on the discrimination characteristics of a detector optimum for an additive mixture of the reflected signal and sea noise. Let us assume in this case that sea noise in the detector bandpass is approximated by white Gaussian noise with spectral density $N_0/2$ [1, 2]. The algorithm of the working principle of discriminating two signals $s_1(t)$ and $s_2(t)$ has the following form [3]:

$$M = \frac{2}{N_0} \int_0^T x(t)[S_1(t) - S_2(t)]\, dt \underset{2}{\overset{1}{\gtrless}} h . \tag{1}$$

Here $h = \ln(P_2/P_1) + (E_1 - E_2)/N$, P_j is the a priori probability of the presence of signals S_j ($j = 1, 2$) and $E_j = \int_0^t s_j^2(t)dt$ reflected from objects c_1 and c_2. The realization arriving at the input of the detecting device (1) in the observation time interval [0, t] is written in this case as $x(t) = S_j(t) + V(t) + h(t)$, where V_t is the reverberation process, which is a Gaussian process with zero mean value and correlation function $B(t_1, t_2) = \langle V(t_1)V(t_2)\rangle$.

Having determined the arbitrary probability densities $W_j(m)$ of the values of M in the presence of the j-th object, let us find the mean probability of the discrimination error for the case typical in radar and sonar with $P_1 = P_2 = 1/2$; $E_1 = E_1 - E$

$$P_e = 1 - \Phi\left[\frac{z(1-R_s)}{\sqrt{2(1-R_s-R)}}\right] . \tag{2}$$

where $\Phi(u) = \frac{1}{\sqrt{2\pi}}\int_0^u e^{-x^2/2}dx$; $R_s = \frac{1}{E}\int s_1(t)s_2(t)dt$ is the cross-correlation coefficient between signals $s_1(t)$ and $s_2(t)$ and $z^2 = 2E/N_0$, $\xi = \frac{1}{N_0 E}\iint B(t_1,t_2)[s_1(t_1)-s_2(t_1)][s_1(t_2)-s_2(t_2)]\,dt_1dt_2$. The derived formula of the mean probability of discrimination error (2) is valid when using any reverberation models, both stationary and transient.

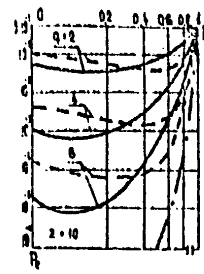

Figure 1

To simplify the derivations, let us consider the behavior of P_e for a specific type of signals, limiting ourselves to the case of stationary reverberation. The latter will be fulfilled [1] on the assumption that long-range reverberation is considered and the reflected signal is processed within a time comparable to its effective length t_d. Using the spectral representations of the signals and reverberation [1], let us analyze the mean probability of discrimination error for the case when objects C_1 and C_2 are located at different distances. The functions $P_e(R_s)$, calculated by the formulas obtained in the paper for the case of a pulsed sounding signal with bell shape (solid lines) $P_e = 1 - \Phi[z(1-R_s)/\sqrt{2-2R_s+\sqrt{2}z^2(1-\sqrt{R_s})/q^2}]$ and of exponential shape (dashed line) $P_e = 1 - \Phi[z(1-R_s)/\sqrt{2\{1-R_s+\frac{z^2}{q^2}[1-R_s(1-\ln R_s)]\}}]$ are plotted in Figure 1. Here z^2 is the signal/noise ratio for power and $q^2 = b_0^2/2\sigma_\xi^2$ is a parameter which characterizes the ratio of intensity b_0^2 of the signal reflected from the field to the dispersion of the reverberation process $V(t)$. The function $P_e^*(R_s)$ in the absence of reverberation $(q \to \infty)$, calculated by the formula [3] $P_e^* = 1 - \Phi[z\sqrt{(1-R_s)/2}]$, is plotted by the dot-dashed line in the same figure for comparison. As follows from Figure 1, the presence of reverberation reduces the efficiency of signal discrimination. In this case the minimum value of P_e, which corresponds to orthogonal signals $(R_s - 0)$ in the absence of reverberation, is reached in its presence at $R_s \neq 0$ and the position of the minimum will depend on the values of q and z. The effect of reverberation on the discrimination characteristics in the presence of randomly moving scatterers (whose velocities [1, 2] are random Gaussian values with zero mean value and mean square deviation of the Doppler reverberation shift Ω) are illustrated for the case of a pulsed bell-shaped sounding signal by the graphs $P_e(R_s)$ in Figure 2. Analysis of the graphs constructed by the derived formula $P_e = 1 - \Phi[z(1-R_s)/\sqrt{2[1-R_s+z^2(1-(R_s)^{1/(1+\xi)})/q^2\sqrt{1+\xi}]}]$, (where $\xi = 1 + 4\Omega^2 t_E^2$) shows that the probability of discrimination error is reduced with an increase of the value of ξ. This is explained by variation of the shape of the spectral density of the mixture of sea and reverberation noise with an increase of the mean square deviation of the Doppler reverberation shift.

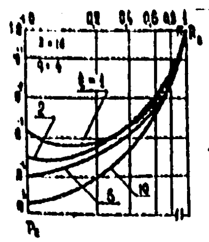

Figure 2

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GEOPHYSICS, ASTRONOMY AND SPACE

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USING THE INVARIANCE PRINCIPLE IN SONAR PROBLEMS WITH A PRIORI AMBIGUITY

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 100-111

[Article by Yu. Ye. Sidorov]

[Text] To overcome parametric and nonparametric a priori ambiguity [1], an effective method in some cases is synthesis based on the use of the invariance principle in the theory of checking complex statistical hypotheses [2]. The use of this method is possible only if the considered statistical problem has symmetry whose mathematical expression is also invariant with respect to a specific group of transformations.

A brief outline of the main propositions of the theory of synthesizing invariant rules of decision-making is presented in the given paper, the main steps of synthesis and the conditions at which construction of optimum invariant rules is possible are enumerated and an example of constructing decision-making rules with respect to sonar problems under conditions of both parametric and nonparametric a priori ambiguity is considered. The main material used in outline of the theoretical propositions is the fundamental monograph [2] (Chapters 1 and 6) and survey papers [3, 4] and monographs [5, 6, 7] were used as auxiliary material.

1. In parametric a priori ambiguity when the functional form of the distribution law of input data is known and ambiguity is expressed in lack of knowledge of the parameters of this law, the problem of synthesizing the rule of selecting the solution (the problem of detection or classification when the number of classes is equal to two*) may be formulated as a problem of finding the rule for checking a complex hypothesis H with respect to a complex alternative K about a multidimensional parameter θ in the general case of some set of distributions $T = \{P_\theta(x), \theta \in \Omega, x \in X\}$, where Ω and X are parametric and selective spaces which are domains of a finite-dimensional Euclidean space, x is the observed sample, $P_\theta(x)$ is the distribution density of the observed example with respect to σ -- a finite measure. The value of parameter θ is then unknown and it is also not assumed that

*Let us consider only these problems for simplicity.

its a priori distribution is given. Only a number of hypotheses on the affiliation of θ to a specific domain of space Ω is formulated. The checked hypotheses have the form: H: $\theta \in \Omega_k$ (there is no signal or a zero-class situation, (K: $\theta \in \Omega_k$)(there is a signal or a first-class situation), where Ω_H and Ω_k are nonintersecting domains of space $\Omega = \Omega_H \cup \cup \Omega_k$, corresponding to hypotheses H and k.*

The theory of synthesizing invariant rules is based on concepts of the symmetry of a set of distributions and symmetry of the problem of checking hypotheses with respect to some class of mutually identical transformations of the parametric space to itself [2].

The set of distributions $T = \{P_0(x), \theta \in \Omega, x \in X\}$ is symmetrical with respect to the class of transformations B if the induced mutually identical transformation $\bar{g}$ of the sampling space X onto itself corresponds to each transformation $g \in B$, which is fulfilled by the equality: $P_{g\theta}(\bar{g}x)I\bar{g} = P_\theta(x)$ at any values of $\theta \in \Omega$ and $x \in X$, where $g\theta = \bar{\theta}'$ and $\bar{g}x = x'$ are the transformed values of parameter θ and samples x and $I\bar{g}$ are Jacobian transforms.

The problem of checking the hypothesis H: $\theta \in \Omega_H$ with alternative K: $\theta \in \Omega_k$ remains symmetrical with respect to class B if the set of distributions T is symmetrical with respect to B and if the loss function $L(\theta, d)$ does not vary during homomorphic mapping of the space of solutions D onto itself, i.e.:

$$L(g\theta, g^*d) = L(\theta, d) \tag{1}$$

at any values of $g \in B$, $\theta \in \Omega$ and $d \in D$, where g* is a transform induced in the space of solutions D and d is a separate solution. In many problems, including detection problems, the loss function $L(\theta, d)$ on sets Ω_H and Ω_k is usually constant and the space of solutions D consists of two elements. Therefore, equality (1) may be written in the form:

$$g\theta \in \Omega_H(\Omega_K)$$

if $\theta \in \Omega_H(\Omega_k)$ at any value of $g \in B$. In other words, if any transformation g from class B retains the parametric set Ω ($g\Omega = \Omega$) and the subsets Ω_H and Ω_k ($g\Omega_H = \Omega_H$ and $g\Omega_k = \Omega_k$), the problem of checking hypotheses K and H is symmetrical (invariant) with respect to g.

With symmetry of the parametric and sampling space expressed by class of transformations B and class of induced transformations $\bar{B}$, it is natural to be limited only by a class of symmetrical (or invariant) rules which solve the functions $\varphi(x)$ of which satisfy the relation:

*In solving the problem of classification with the number of classes greater than two, the hypothesis H is checked with respect to many alternatives K: $\theta \in \Omega_{k_i}$ (situations of i-th class), i = 1, 2, ..., n and $\Omega_H \cup \Omega_{k_i} = \Omega$.

$$\varphi(\bar{g}x) = \varphi(x) \qquad (2)$$

for all values of $\bar{g} \in B$ and $x \in X$.

The power functions of invariant rules determined by the conditions $\beta_\varphi(\theta) = E_\theta[\varphi/x]$, where $E_\theta[\cdot]$ is a sign of averaging not dependent on variations of parameter θ due to the effect of transformations of class B, which as will be obvious from the examples, is their practically valuable property.

An important step of synthesizing invariant rules is to find a special function ($z(x)$ in the sampling space and $v(\theta)$ in the parametric space), called the maximum invariant (MI) by which all the invariant functions are expressed and, specifically, by which the resolving and power functions are expressed. We note that the function $z(x)$ (similar to $v(\theta)$), is called an MI if:

1) it is invariant, i.e., $z(\bar{g}x) = z(x)$ at any value of

$$\bar{g} \in \bar{G}. \qquad (3)$$

and if 2) the condition is fulfilled: it follows from $z(x') = z(x'')$ that

$$x'' = \bar{g}x' \qquad (4)$$

at some value of $\bar{g} \in \bar{G}$.*

The essence of this step consists in the fact that a set of equivalent points called trajectories (or orbits) $\bar{G}$ in the sampling space and which form its division is found with sequential application of all transformations $\bar{g}$ from $\bar{G}$ to $x \in X$. Point x "passes through" the trajectory when all the transformations $\bar{g}$ from $\bar{G}$ are applied to it. This means that the trajectory containing this point consists of all points $\bar{g}x$ from $\bar{g} \in \bar{G}$. In these terms $z(x)$ or $v(\theta)$ is the MI if it is constant on each trajectory (condition (3)) and if it assumes different values on different trajectories (condition (4)). All the MI are equivalent in the sense that the sets of points where they are constant coincide.

After finding the MI, the initial problem is considerably reduced, i.e., it is reduced to new random functions $z(x)$ and $v(\theta)$, on the basis of which all the invariant rules are constructed. The following relations are valid for these rules: $\varphi(x) = \psi[z(x)]$ and $\beta_\varphi(\theta) = F_\varphi[v(\theta)]$. These relations permit one to turn from the initial set of distributions $\{P_\theta(x)$, $\theta \in \Omega$ and $x \in X$ to the set of distributions of the MI $\{P_v(z)$, $v \in V$ and

*Classes of transformations B and $\bar{B}$ may always be expanded to groups G and $\bar{G}$, the operation determinant in which is sequential use of two transformations (multiplication of transformations), which facilitates finding the MI.

$z \in Z$, where V is the domain of values of the function $v(\theta)$ and Z is the domain of values of function $z(x)$. Thus, invariance by reduction of the data to the MI of $z(x)$ and $v(\theta)$ reduces both the sampling and parametric spaces. It is significant that the distribution $P_{v(\theta)}[z(x)]$ depends only on the MI of $v(\theta)$ in the parametric space.

Thus, the main steps of synthesizing invariant rules are:

1. Determining the groups G and $\overline{G}$ of transformations $\overline{g}$ and g of the parametric and sampling spaces which retain the invariant problem of checking the hypothesis H with respect to the alternative K.

2. Finding the MI of $z(x)$ and $v(\theta)$ with respect to groups $\overline{G}$ and G.

3. Constructing the resolving function $\psi[z(x)]$ and the power function $F_{\varphi}[v(\theta)]$.

Since the final goal of synthesis is to find the optimum (by any criterion) resolving rule, the uniformly most powerful (RNM) rules having maximum probability of correct solution at any values of a priori unknown parameter $\theta \in \Omega_k$ is of greatest interest among the invariant rules. Unfortunately, the existence of the RNM of rules generally and of the RNM of invariant rules (RNMN) specifically may not always be shown. Additional requirements must be fulfilled to accomplish this in order that the domain of values of MI of $z(x)$ be a one-dimensional space. A more significant and more important requirement that the set of distributions of MI $\{P_v(z)\}$ have monotonic similarity $P_{v'}(z)/P_{v''}(z)$ with respect to z, i.e., so that the distributions $P_{v'}$ and $P_{v''}$ be different for any values of v' and v" and that their ratio $P_{v'}(z)/P_{v''}(z)$ is a non-decreasing function of MI $z(x)$.

If all the requirements are fulfilled, the optimum (by the most widespread Neyman-Pearson criterion in detection and classification) are the resolving functions of RNMI rules provided that the domain of values of the MI in the parametric space $v(\theta)$ is a one-dimensional space or that a fixed value of $v(\theta) = v_0$ corresponds to hypothesis H, may be written in the form:

$$\psi[z(x)] = \begin{cases} 1 & \text{at } z(x) > C, \\ \gamma & \text{at } z(x) = C, \\ 0 & \text{at } z(x) < C, \end{cases} \tag{5}$$

where threshold C and probability γ are selected so as to fulfill the equality

$$E_{v_0}[\psi(z)] = \alpha. \tag{6}$$

Here α is the level of significance (the probability of a false alarm or the probability of false classification), v_0 is the value of $v(\theta)$ on the set Ω_H or on the boundary Ω_H and Ω_k and $E_{v_a}[\cdot]$ is the sign of averaging. The power function $\beta_\varphi(\theta) = F_v[v(\theta)] = E_\theta[\varphi(x)]$ of rule (5) increases strictly at all points where $\beta_\varphi(\theta) < 1$.

As can be seen from (5), the rule does not depend on the unknown parameters of the set of distributions $T = \{P_\theta(x), \theta \in \Omega, x \in X\}$. This indicates that the device realizing this rule will be invariant to any actual values of the observed distribution parameters. Moreover, as can be seen from (6), the rule has constant probability of a false solution at any values of θ (the value of v_0 belongs to the boundary Ω_H and Ω_k or is given by hypothesis H; therefore, it is a priori known), which does away with the requirement of adjusting the threshold during operation of the device for the actual values of the distribution parameters of the observed sample. These valuable properties permit automation of the decision-making procedure (detection and classification). Consequently, the principle of invariance made it possible to exclude the dependence of the probabilities of solution from a previously established class of variations of distribution parameter.

If condition (2) is written in the form $\varphi(\bar{g}_L x) = \varphi(x)$ for all values of $x \in X \setminus N_{\bar{g}}$, $\bar{g} \in \bar{G}$, where the exclusive set $N\bar{g}$ of measure zero possibly depends on $\bar{g}$, function φ is called almost invariant with respect to group $\bar{g}$. This definition is used when studying the relationship between invariance on the one hand and nonbias and some other optimum properties on the other. It also helps in answering the question of what application of the invariance principle to the power function yields. It turns out that almost invariance with respect to group G of the resolving rule (resolving function) φ and invariance with respect to its power function become equivalent if the problem is reduced to adequate statistics T(x), the distribution of which form a limited complete set, prior to application of the invariance principle. It is shown in [2] with the existence of these statistics and with rather general assumptions about the group of transformations G that the RNMI rules are also RNM in a broader class of rules with invariant power functions. This of course assumes proof (under specific, rather broad conditions) of the equivalence of RNMI and RNM of almost invariant rules, which was also done in [2].

Let us combine all conditions which provide synthesis of RNMI rules.

1. The problem of checking the hypotheses has symmetry with respect to some class of one-two-one transformations of the parametric space onto itself.

2. The range of values of MI z(x) is a one-dimensional space.

3. The set of MI distributions has monotone ratio of similarity with respect to z.

If in addition to these three conditions, one can prove the existence of adequate statistics with bounded complete set of distributions, the rule will be RNM not only in the class of invariant rules but also in the broader class of rules with invariant power functions.

II. With nonparametric a priori ambiguity, the functional form of the distribution law of observed data itself is unknown and the problem of synthesizing the rule of decision-making is formulated as a problem of checking hypotheses H and K on the differences between distributions. Moreover, this difference between hypotheses is given in some general form unrelated to a specific type of distribution function. It is of interest to systematize the main nonparametric problems of checking the hypotheses, for solution of which the invariance principle may be used. This is necessary and useful to illustrate how the hypotheses and the difference between them are given since, unlike the parametric case, hypotheses H and K are written differently here for each type of problem.

1. The problem of conformity. Let the results of observations form two samples $x_1, \ldots, x_m$ and $y_1, \ldots, y_n$ from distributions F_1 and F_2 which belong to a class of all continuous distribution functions. The competing complex hypotheses H and K may be written thusly: $H:F_1 = F_2$; $K':F_2(z) \leqslant F_1(z)$ for all values of z and $F_2 \neq F_1$:$K'':F_2(z) \geqslant F_1(z)$ for all values of z and $F_2 \neq F_1$; $K''':F_1 \neq F_2$. The alternative K' denotes that the values of Y^2 are stochastically greater than the value of X and that K' denotes that the values of X are stochastically greater than the values of Y ($X = (x_1, \ldots, x_m)$, $Y = (y_1, \ldots, y_n)$).

In the simplest case, distribution F_1 may be known and F_2 may be unknown. Hypothesis H will then be simple and the complex alternatives will be written in similar fashion.

2. The problem of symmetry. If the distributions of independent observations $(x_1, y_1), \ldots, (x_N y_N)$ are described by a continuous distribution function F, then function F is symmetrical with respect to the line $y = x$ if $H:F(x, y) = F(y, x)$ and is asymmetrical if $K:F(x, y) \neq F(y, x)$.

The symmetry of function F(x) with respect to point x_0 is checked by the hypotheses: $H':F^{-1}(0.5) = x_0$ (F(x) is symmetrical) and $K':F^{-1}(0.5) \neq x_0$ (F(x) is asymmetrical).

3. The problem of independence. If $(x_1, y_1), \ldots, (x_N, y_N)$ is a sample of two-dimensional distribution F, then independence X and Y is checked by the hypothesis $H:F(x, y) = F_1(x) \cdot F_2(y)$ with respect to the alternative $K:F(x, y) \neq F_1(x) \cdot F_2(y)$.

The number of multipliers in the hypotheses will naturally be K for a K-sample problem $k = 1, 2, \ldots, k_0$.

A positive dependence between X and Y may be suggested as the alternative of K. This formulation occurs when the values of x are not random, but are selected previously, for example, so that $x_1 < \ldots < x_N$. If F_i denotes the distribution Y with given value of x_i, y_i, their distribution functions $F_1, \ldots, F_N$, $H:F_1 = \ldots = F_N$ are also independent. The alternative hypothesis of the positive function y_i consists in a stochastic increase of them with an increase of i: $K':F_1 < F_2 < \ldots < F_N$.

4. The problem of randomness. Let there be a series of h observations of one random value ordered in some manner (for example, in time). Hypothesis K consists in the fact that the observations were obtained independently from each other and from the same general aggregate, i.e., $H: F_1(x) = F_x(x) = \ldots = F_N(x)$ for all values of x. The alternative K (like K' in the problem of independence) indicates a positive trend, i.e., the stochastic increase of x_i with an increase of i: $K: F_1(x) < F_2(x) < \ldots < F_n(x)$ for all values of x, where the values of x_i remain independent.

This problem is equivalent to checking the independence of x and y, but the accent is made here on the identical distribution of the values of x_i.

5. The problem of displacement. If the factor of interest to us leads to a shift of distribution F(x) in any direction (it is unknown in which direction), the problem of detecting the shift Δ, also called the problem of arrangement or localization, occurs, i.e., we have an alternative of type: $F_1(x) = F_2(x - \Delta)$. The hypotheses may be formulated thusly: $H: \Delta = 0; \Delta > 0; K'': \Delta < 0; K''': \Delta \neq 0$.

6. The problem of scale. With variation only of the distribution scale, we have an alternative of type $F_1(x) = F_2(\ \ x)$ and the following possible hypotheses:

$$H: \mu = 1\ (\mu = \mu_0);\ K': \mu > 1\ (\mu > \mu_0);$$
$$K'': \mu < 1\ (\mu < \mu_0);\ K''': \mu \neq 0\ (\mu \neq \mu_0)$$

The enumerated problems are the main and most typical (specifically for sonar) nonparametric problems, for solution of which the invariance principle may be applied, which leads to the best (in a definite sense) procedures of solution. However, they of course do not exhaust all the possibilities of discriminating the nonparametric hypotheses. Additional variants (or modifications) of the enumerated problems can be found in [2, 4, 5, 6].

Each of the problems formulated above has its own group of transformations G of the sample space and its own MI. Without dwelling on each of the hypotheses formulated in these problems, let us note only those transformations $\bar{g} \in \bar{G}$ which lead to synthesis of extremely interesting and practically valuable rank rules of decision-making [2, 1, 5, 6] which are attracting ever greater attention of developers of detection systems and classification of their own efficiency and effectiveness in nonparametric a priori ambiguity with each year.

1. In the problem of conformity, the problem of checking hypothesis H with alternative $\bar{K}$ remains invariant with respect to group G of all transformations $x_i' = \bar{g}(x_i)$, $y_j' = \bar{g}(y_j)$ ($i = 1, \ldots, m$; $j = 1, \ldots, n$) such that the functions are continuous and strictly increase. This statement is possible since these distributions retain continuity of distributions and the properties of the value remain identically distributed or one of them will be

stochastically greater than the other. The MI with respect to group $\bar{G}$ is a set of ranks $(R'; S') = (R_1, \ldots, R_m; S_1, \ldots, S_n)$ of the values of $x_1, \ldots, x_m$ and $y_1, \ldots, y_n$ considered as elements of a single sample.

2. It is convenient to make the transformation $z_i = y_i - x_i$ and $w_i = x_i + y_i$ [2] in the problem of symmetry for implementing this reduction for hypothesis H. Then H consists in the fact that continuous two-dimensional distribution $(z_i; w_i)$ is symmetrical with respect to axis w and with the alternative this distribution is shifted in the positive direction of axis z. This problem is invariant with respect to one-to-one transformations $w' = \bar{g}_1(w_i)$, where $\bar{g}_1$ has its own more finite number of separation points. The MI with respect to this group will be the function $(z_1, \ldots, z_N)$. The hypothesis of symmetry of the continuous one-dimensional distribution function F of values z_i with respect to the origin $H': F(z) + F(-z) = 1$ for all values of z is checked with the alternative at F is shifted in the positive direction of the z-axis. This altered problem is invariant with respect to group G of all transformations $z_i' = \bar{g}_1(z_i)$, $i = 1, \ldots, N$, where $\bar{g}_1$ is continuous, odd and strictly increases. The MI are ranks $s_1', \ldots, s_n'$ of the values of $z_{j1}, \ldots, z_{j,n}$ among the absolute values $z_1, \ldots, z_N$ and the ranks $r_1, \ldots, r_m$ of values $z_{i_1}, \ldots, z_{i_m}$ among $z_1, \ldots, z_N$.

3. In the problem of independence the problem of checking hypothesis H with the alternative of the positive dependence is invariant to transformations $x_i' = \bar{g}_3(x_i)$, $y_i = \bar{g}_4(y_i)$ so that $\bar{g}_3$ and $\bar{g}_4$ are continuous and strictly increase. The MI with respect to these transformations are ranks $(R_1', \ldots, R_N')$ of values $(x_1, \ldots, x_N)$ among X and the ranks $(S_1', \ldots, S_N')$ of values $(y_1, \ldots, y_N)$ among Y.

4. In the problem of randomness, the problem of checking H with K is invariant with respect to the group of transformations $x_i' = f(x_i)$, where function f is continuous and strictly decreases. The MI is the set of ranks $(S_1, \ldots, S_N)$ of values $(x_1, \ldots, x_n)$.

A very valuable property of rank rules, included in their stability with respect to variation of the distribution law of observed data with zero hypothesis H (in detection this hypothesis indicates the absence of a shaft and in classification it indicates affiliation to the zero class), was noted in [1]. Moreover, rank procedures are accomplished simply and rapidly and they may be used when only the results of ordering the observations are known. The priority of rank rules is also determined by their high asymptotic effectiveness, which is hardly inferior to the potential effectiveness of Bayes rules which utilize complete a priori information about the statistics of signals and noise. It is now known [5] that at least in some cases the results of the theory of rank criteria may be used to find the asymptotically optimum criteria in the class of all criteria.

However, the range of application of rank procedures has natural boundaries [5]. Rank procedures are appropriate only for simple experiment plans with sufficiently large groups of identically distributed (with zero hypothesis) observations. In the final samples, the effectiveness of rank criteria is lower than that of the best parametric criteria, which is payment for the high stability of rank rules in the absence of a priori data. Therefore, it is natural that, for example, the probability of correct detection of a signal in rank rules will be less than the maximum theoretically possible and will depend on the noise distribution in this case. However, losses in the effectiveness of rank rules are usually low if only the level of significance is not very small and, moreover, the rank rules frequently utilize information about the type of distribution more flexibly than parametric rules.

The best known and most widely used among rank rules are;* a) the Wilcoxson rule (Manna-Whitney), according to which a decision is made in favor of hypothesis K if $\sum_{i=1}^{R} R_i \geqslant c$, where $R_1, \ldots, R_N$ are components of the rank vector and [translator's note: one letter missing] is the threshold number determined by the probability of error of first kind; b) the Fischer-Yates-Terry-Hefding rule with range of rejection of the zero hypothesis $\sum_{i=1}^{N} E(V^{(i)}) \geqslant$ $\geqslant c$, where E is a sign of averaging which occurs by distribution of the ordered sample $V^{(1)} < \ldots < V^{(N)}$ of volume N from the standard normal distribution; and c (the Van den Varden rule with range of rejection of the zero hypothesis $\sum \Phi^{-1} (R_i/(N + 1)) \geqslant c$, where ϕ^{-1} is a function inverse to $\phi(x) = (2\pi)^{-1/2} \int_{-\infty}^{x} e^{-t/2}dt$.

There are presently no clear recommendations for construction of RNM of rank rules (as it is still generally impossible to hope to find nonparametric rules which would be more powerful opposed to all possible alternatives). These rules may most frequently be found opposed to a limited class of normal alternatives (and then not always). Therefore, let us fulfill the main required conditions here at which synthesis of RNMI rank rules may be accomplished.

1. The problem of checking the hypotheses is invariant with respect to any monotone transformations of the sample space which retain distribution continuity and the properties of the observed values will be distributed according to one or another stochastic ordering, for example, they will be identically or differently distributed.

2. The MI $\phi(R)$ is one-dimensional.

---

*The rules given below are linear since they are based on linear rank statistics of type $\sum a(i, R_i)$, where $\{a(i, j)\}$ is the arbitrary matrix N x N, unlike rank rules based on Kendall, Spirmen, Kolmogorov-Smirnov, Sen'i, Kramer-Mises and other statistics which are not linear.

3. The set of MI distributions has monotone plausibility ratio with respect to R.

These conditions are apparently valid for synthesis of other nonparametric rules as well, not only rank rules (of course, with the appropriate refinements).

The power function of the RNMI rank rule is constructed for a specific parametric alternative (with given type of distribution). As in section 1, if it is possible to show the existence of adequate statistics with limited complete set of distributions when fulfilling all three conditions, the rank rule will also be RNMI in the class of rules with invariant power functions.

It is most frequently possible to show (especially in the problem of displacement) the existence of locally more powerful (LNM) rules.

Conclusions. The use of the invariance principle in sonar problems about the different level of a priori ambiguity permits one to solve the timely problem of synthesizing dependable and effective algorithms (and the corresponding devices) for primary and secondary processing of sonar information, efficient under conditions of a lack of a priori data and comparatively simple to realize by using digital techniques (digital computers). Proof of this is the considered examples (and others published previously). The invariant resolving rules found in them do not depend on a priori unknown characteristics (parameters) of observed data (they have structural stability) and their power functions have the desirable properties of invariance with respect to the interfering effect of unknown parameters (stability of quality). Therefore, one may rightfully talk about the constructiveness of this method of solving various sonar problems.

It is natural that the use of invariance is in no way universal, but is limited by a class of symmetrical problems. Moreover, as indicated in [1], it is not completely algorithmic (there are no clear recipes for finding groups of transformations and also minimum invariants for a number of problems). However, a considerable reduction of observed data (i.e., discarding of unnecessary information) can be accomplished by using it and thus the dimensionality of the initial space can be reduced, which is extremely important for reducing the load on the digital computer memory; it permits one to turn very elegantly to rank rules, whose role in overcoming nonparametric a priori ambiguity is rather high and, which is most important, its constructiveness permits one to synthesize devices with a desirable property of invariance to a priori unknown characteristics of observed processes. All this permits one to evaluate the considered approach as promising (but of course not the only one) for developing automatic digital sonar data processing systems under conditions of a priori ambiguity.

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GEOPHYSICS, ASTRONOMY AND SPACE

UDC 621.391.2

ADAPTIVE DETECTION OF NOISY SIGNALS

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 112-116

[Article by L. G. Krasnyy and V. P. Peshkov]

[Text] Two approaches to the synthesis of adaptive processing algorithms are possible according to the results of [1-2]. According to one of them, the optimum adaptive detector is constructed on the basis of a traditional optimum detector (ratio of similarity) in which the estimates obtained during training are substituted instead of the unknown characteristics of signals and noise.

Application of this methodology to the problem of detecting a Gaussian signal on a background of Gaussian noise leads to the following adaptive algorithm:

$$\mathcal{F}(U) = \int_{-\infty}^{\infty} \left\{ \frac{1}{g_N^*(\omega)} - \frac{1}{g_{SN}^*(\omega)} \right\} |V(\omega)|^2 d\omega, \tag{1}$$

where $V(\omega)$ is a Fourier transform of adopted realization $U(t)$, $g_N^*(\omega)$ and $g_{SN}^*(\omega)$ are estimates of the spectral noise densities ($g_N(\omega)$) and a mixture of signal and noise ($y_{SN}(\omega) = g_S(\omega) + g_N(\omega)$).

Algorithm (1) describes a typical "filter-detector-filter" detection channel with adaptive filter-preselector whose frequency characteristic $(K(\omega))^2 = 1/g_N^*(\omega) - 1/g_{SN}^*(\omega)$ is rearranged during the teaching process. Technical realization of this preselector is possible by using the comb of narrowband filters (or BPF [expansion unknown] processor) whose outputs are combined with weights $(K(\omega_i))^2$. The estimates $g_N''(\omega)$ and $g_{SN}''(\omega)$ required to select the weighting coefficients can be found on the same comb of filters.

In the proposed adaptive detector, the spectral density $g_N(\omega)$ is estimated in the reference channel (for example, in an adjacent spatial channel) in which the signal is a priori absent, while estimation of $g_{SN}(\omega)$ is made in the working channel, i.e., by unclassified sampling.

Quantitative investigation of the noise stability of an adaptive detector (1) was carried out by statistical modeling methods on the M-222 computer. Some results of these investigations are presented in Figure 1. Gaussian noise with spectral density $g_S(\omega) = g_S d^2/(d^2 + \omega^2)$ ($d = 2\Delta f_s$, Δf_s is the signal band) was used as the signal model and white noise with spectral density g_N was used as noise. It follows from the figure that even with relatively short measurement time T_i, the adaptive detector is slightly inferior (approximately 1.5 dB at D = 0.9) to the optimum detector, remaining 3 dB better than a detector with nonoptimum preselector.

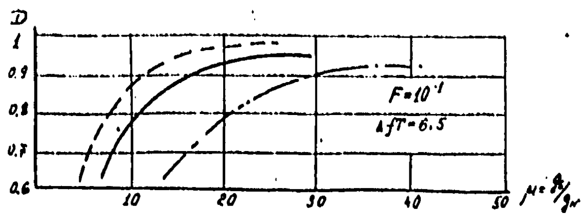

Figure 1. Characteristics of Detecting an Adaptive Algorithm (1) (Solid Curve), Optimum Detector (Dashed Curve) and Channel With Ideal Band Filter (Dot-Dashed Curve)

Let us now consider another method of synthesizing adaptive detectors, based on the theory of $\mathcal{E}$-optimization of data processing systems [2]. According to this theory, the optimum algorithm for detecting a noisy signal s(t) with arbitrary distribution on a background of non-Gaussian N(t) and white n(t) noise in the most general form may be described by the stochastic differential equation:

$$d\mathcal{F}_t(u) = \mathcal{F}_t(u)[1-\mathcal{F}_t(u)][s^*(t)+N_1^*(t)-N_0^*(t)]\{u(t)- -N_0^*(t)-\mathcal{F}_t(u)[s^*(t)+N_1^*(t)]\}dt; \quad \mathcal{F}_{t=0}(u)=\xi, \tag{2}$$

where $F_t(u)$ is the $\mathcal{E}$-optimum processing functional, $s^*(t)$, $N_1^*(t)$ and $N_0^*(t)$ are the optimum estimates (by the mean square criterion) of the signal S(t) and noise N(t) and the estimate $N_1^*(\tau)$ is calculated in the presence of a signal and $N_0^*(t)$ is calculated in its absence.

The main difficulty in realization of algorithm (2) is optimization of the signal and noise filter blocks, which requires their complete probability description. One of the effective methods of overcoming this difficulty is description of Gaussian signals and noise by differential equations of type:

$$\frac{d^m s(t)}{dt^m} + \sum_{k=0}^{m-1} \alpha_k \frac{d^k s(t)}{dt^k} = \sqrt{\frac{g_s}{2}}\,\frac{dw_s(t)}{dt} \tag{3}$$

where $w_s(t)$ is a standard Wiener process.

With regard to (3), synthesis of the optimum filtration block reduces to the following: equation (3) is supplemented by trivial relations $d\alpha_k = 0$ for unknown weighting coefficients and the results of optimum nonlinear filtration theory are applied to the derived system of equations. As a result we arrive at a system of stochastic differential equations which describe the structure of the optimum filtration block. It is obvious that the complexity of this block is determined by the order of equation (3), an increase of which permits one to encompass a rather wide class of signals with practically arbitrary densities.

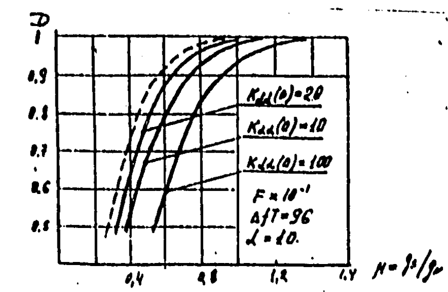

Figure 2. Characteristics of Detecting Adaptive Processing Algorithm (2) and (5)

However, one can show that an extreme increase of "m" in (3) is not feasible. It turns out that even the use of the simplest representations:

$$dS(t) = -\alpha S(t)\,dt + \alpha\sqrt{\frac{g_s}{2}}\,dw_s(t) \qquad (4)$$

with optimum selection of α permits one to provide noise stability close to potential. This circumstance considerably simplifies the optimum filtration block. In this case the system of equations which describes its structure assumes the form (for simplicity of outlining it is suggested that $N(t) = 0$):

$$\begin{cases} \dfrac{dS^*(t)}{dt} = -\alpha^*(t)S^*(t) - K_{\alpha s}(t) + \dfrac{2}{g_N} K_{ss}(t)[U(t) - S^*(t)], \\ \dfrac{d\alpha^*(t)}{dt} = \dfrac{2}{g_N} K_{\alpha s}(t)[U(t) - S^*(t)], \\ \dfrac{dK_{ss}(t)}{dt} = -\dfrac{2}{g_N} K_{ss}^2(t) - 2\alpha^*(t)K_{ss}(t) - 2S^*(t)K_{\alpha s}(t) + \\ \qquad + \dfrac{g_s}{2}(\alpha^*(t))^2 + \dfrac{g_s}{2} K_{\alpha\alpha}(t), \\ \dfrac{dK_{\alpha s}(t)}{dt} = -S^*(t)K_{\alpha\alpha}(t) - \alpha^*(t)K_{\alpha s}(t) - \dfrac{2}{g_N} K_{ss}(t)K_{\alpha s}(t), \\ \dfrac{dK_{\alpha\alpha}(t)}{dt} = -\dfrac{2}{g_N} K_{\alpha s}^2(t). \end{cases} \qquad (5)$$

The system of equations (5) jointly with algorithm (2) describes the structure of an adaptive noisy signal detector with a priori unknown spectral density. The characteristic feature of the synthesized detector is that, unlike (1), there is no time separation of the teaching and detecting stages here and estimation and detection are carried out in the same time interval.

Algorithms (2) and (5) were modeled on the M-222 computer, which permitted experimental investigation of its noise stability. The detection characteristics of this algorithm are shown in Figure 2 by the solid lines and the dashed line corresponds to the optimum detector.

The process of establishing parameter α as a result of its filtration according to algorithm (5) is shown in Figure 3.

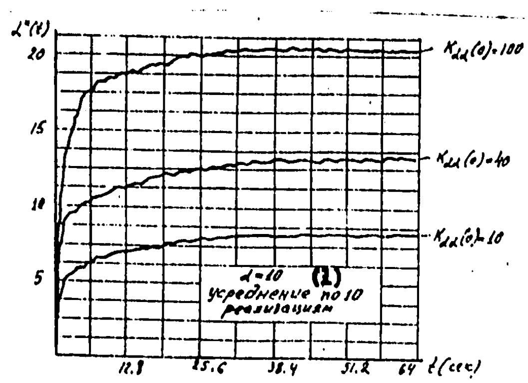

Figure 3. Process of Establishing Parameter α During Adaptation by a Given Parameter

KEY: 1. Averaging by 10 realizations

The a priori dispersion $K_{\alpha\alpha}(0)$ of coefficient α (the a priori mean value of α was assumed equal to zero and the true value of $\alpha_0 = 10$) was selected as the parameter in Figures 2 and 3. It is obvious that the steady value of parameter α and consequently the noise stability of an adaptive algorithm depends on the selected a priori dispersion $K_{\alpha\alpha}(0)$ of coefficient α. In this regard a special procedure of selecting $K_{\alpha\alpha}(0)$ becomes necessary, but this problem is the subject of separate investigations.

The methodology of synthesizing adaptive detectors, described above, may also be applied to more complex situations, including cases of narrowband signals and noise.

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GEOPHYSICS, ASTRONOMY AND SPACE

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SECONDARY PROCESSING OF MULTIBEAM SIGNALS UNDER CONDITIONS OF INCOMPLETE A PRIORI INFORMATION

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 117-121

[Article by N. G. Gatkin, L. N. Kovalenko, L. G. Krasnyy and S. V. Pasechnyy]

[Text] The algorithm of optimum detection of multibeam signals consists of two sequential procedures: primary and secondary processing. Primary processing is accomplished by using correlators or optimum filters and serves for optimum processing of signals propagated along separate beams. Secondary processing is used to improve noise stability by taking into account the structure of the multibeam signal. In the general case it includes cross-correlation processing of output voltages $U(t - \tau_i)$ of the primary processing channel, taken at moments of time corresponding to the anticipated delays τ_i of the multibeam signal [1]. If signals propagated along different beams can be resolved as a result of using complex sounding pulses after primary processing, secondary processing is considerably simplified and consists in storage of voltages $U(t - \tau_i)$. Thus, optimization of secondary processing requires a priori information about the number of beams and their delays with respect to each other in all cases. In some cases this information can be found from the results of predicting the refraction pattern. Nevertheless, situations are possible when the dependability of data obtained during hydrological calculations is low. In this case an essentially different postulation of the optimization problem is required -- optimization under conditions of incomplete a priori information.

To solve this problem, let us use the theory of ε-optimization of data processing systems [2] which makes it possible to overcome the "a priori difficulty" indicated above. Let us reformulate the problem of secondary processing in the following manner. Let the output voltage U(t) of the primary processing channel be quantified in time with step $\Delta\tau$, equal to the range resolution of the system (for example, when using LChM [Linear Frequency Modulation] of the signal, $\Delta\tau = 1/\Delta F_g$, where ΔF_g is frequency deviation). Then

$$U(i\Delta\tau) = U_i = S_i + N_i , \quad (1)$$

where S_i and N_i are the signal and noise, respectively, at the output of the primary processing channel.

If the checked hypothesis consists in determination of the presence of a target at range r_k, corresponding to the arrival time of the signal $\tau_k = k\Delta\tau$, then voltage U(t) is observed on the interval $[\tau_k, \tau_k + \tau_{max}]$, where τ_{max} is the maximum possible delay of a multibeam signal. Accordingly, U_i is observed at $i = k, k + n$, where $n = \tau_{max}/\Delta\tau$.

Since neither the number of beams nor their delays are a priori known, signal S_i in (1) is a discrete random process with unknown distribution. Calculations of N_i of noise are uncorrelated since the correlation interval of noise and reverberation noise does not exceed the quantification step in time $\Delta\tau$.

The algorithm of $\mathcal{E}$-optimum detection of a random signal S(t) with arbitrary distribution on a background of uncorrelated noise has the form [2]:

$$U_t = \int_t^{t+T} U(\tau) S^*(\tau) d\tau - \frac{1}{2} \int_t^{t+T} [S^*(\tau)]^2 d\tau, \tag{2}$$

where S*(t) is the optimum (by the mean square criterion) estimate of signal S(t).

Consequently, going from (2) to the case of discrete time, we find the algorithm for quasi-optimum processing of signal S_i:

$$U_k = \frac{1}{n} \sum_{i=k}^{k+n} U_i S_i^* - \frac{1}{2n} \sum_{i=k}^{k+n} (S_i^*)^2. \tag{3}$$

To construct the estimate of S_i^*, let us represent S_i in the form:

$$S_i = \theta_i a_i\ , \tag{4}$$

where θ_i is a random value which assumes values "0" or "1," and a_i are weighting coefficients corresponding to the signal level on the i-th segment of range. This representation characterizes the possibility of the appearance or absence of an echo-signal (which may consist of several pips), arriving along the i-th beam with delay $\tau_i = i\Delta\tau$.

The value $S_i^* = \theta_i^* a_i$ follows from (4), hence:

$$U_k = \frac{1}{n} \sum_{i=k}^{k+n} U_i \theta_i^* a_i - \frac{1}{2n} \sum_{i=k}^{k+n} a_i^2 (\theta_i^*)^2. \tag{5}$$

Let us assume that the arrival of the signal along the i-th beam does not depend on the fact of arrival of the signal along the k-th beam ($k < i$). Then, using the result of [2], we find the following expression for the optimum estimate of θ_i^*:

$$\theta_i^* = \frac{\frac{P_i}{1-P_i}\lambda(U_i)}{1+\frac{P_i}{1-P_i}\lambda(U_i)}, \tag{6}$$

where $\lambda(U_i) = (W(U_i - a_i)/W(U_i)$ is the ratio of plausibility, calculated by the known noise distribution $W(U_i)$ for the i-th sample and P_i is the probability of a signal appearing in the i-th beam.

With regard to (6), the algorithm of quasi-optimum secondary processing of a multibeam signal has the form:

$$U_k = \frac{1}{n}\sum_{i=k}^{k+n} a_i U_i \left\{\frac{\frac{P_i}{1-P_i}\lambda(U_i)}{1+\frac{P_i}{1-P_i}\lambda(U_i)}\right\} - \frac{1}{2n}\sum_{i=k}^{k+n} a_i^2 \left\{\frac{\frac{P_i}{1-P_i}\lambda(U_i)}{1+\frac{P_i}{1-P_i}\lambda(U_i)}\right\}^2. \tag{7}$$

The synthesized secondary processing algorithm describes the procedure of weight accumulation of the output effects of the primary processing channel, and the weights are determined automatically from the results of analyzing the sample used. Moreover, algorithms (7) permit the use of a priori information (formulated in probabilistic interpretation P_i) on the presence or absence of a signal propagated along each of the resolved beams. If the fact of the absence of an i-th signal is confidently known ($P_i = 0$), then $\theta_i^* = 0$ and voltage U_i is not accumulated; in the opposite case ($P_i = 1$), U_i is accumulated with maximum weight $\theta_i^* = 1$.

In the absence of a priori data on P_i and a_i, i.e., at $P_i = 1/2$ and $a_i = 1$, instead of (7), we find

$$U_k = \frac{1}{n}\sum_{i=k}^{k+n} U_i \left\{\frac{\lambda(U_i)}{1+\lambda(U_i)}\right\} - \frac{1}{2n}\sum_{i=k}^{k+n} \left\{\frac{\lambda(U_i)}{1+\lambda(U_i)}\right\}^2. \tag{8}$$

Algorithm (8) may be given even greater flexibility if it is replaced by a multichannel system with p stores of different "memory." In this case each channel has the structure:

$$U_{kr} = \frac{1}{r}\sum_{i=k}^{k+r} U_i \left\{\frac{\lambda(U_i)}{1+\lambda(U_i)}\right\} - \frac{1}{2r}\sum_{i=k}^{k+r} \left\{\frac{\lambda(U_i)}{1+\lambda(U_i)}\right\}^2, \quad r = \overline{1,n}, \tag{9}$$

and the output effects u_{kr} are then combined into the scheme of a sample by the maximum (Figure 1).

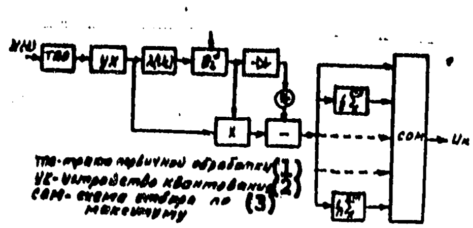

Figure 1

KEY:

1. TPO -- primary processing channel
2. UK -- quantification device
3. SOM -- diagram of sampling by maximum

The effectiveness of the synthesized algorithm for secondary processing of multibeam signals was investigated by modeling with subsequent experimental checking (for 1,000 experiments) of its noise stability. It was assumed during modeling that noise distribution was Gaussian with zero mean value and dispersion $\sigma^2 = 1$, $P_i = 1/2$, $a_i = A = \text{const}$ and $n = 50$ (which corresponds to the maximum delay $\tau_{max} = 500$ msec and LChM signal with deviation $\Delta f_g = 100$ Hz).

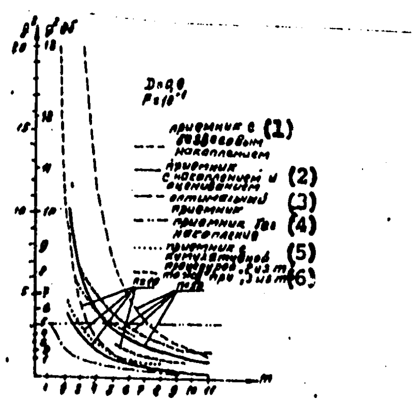

Figure 2

KEY:

1. Detector with unweighted accumulation
2. Detector with accumulation and estimation
3. Optimum detector
4. Detector without accumulation
5. Detector with cumulative procedures "2 and 3 m"
6. Detector with cumulative procedures at "3 and 2 m"

The dependence of threshold signal/noise ratios g^2 (at $D = 0.9$ and $F = 10^{-1}$) on the number of beams (pips) is shown by the solid lines in Figure 2. As can be seen, a detector with accumulation and estimation (9) is 6-7 dB better in noise resistance than an algorithm with unweighted accumulation (dashed lines) with small number of beams ($m/n \ll 1$) and sampling volume $n = 50$. With an increase in the number of beams ($m/\sqrt{n} > 1$), this advantage decreases and comprises 3-4 dB. The advantage also decreases at $n = 10$, i.e., with relatively small sampling volume, and comprises approximately 3 dB at $m/n < 0.3$ and approximately 1 dB at $m/\sqrt{n} \gg 0.3$.

The dot-dashed line shows the dependence of the threshold signal/noise ratio on m for an optimum detector synthesized for the presence of complete a priori information about the number of beams and their delays (this detector stores only those values of U_i in which there is a signal). The line with double dots, parallel to the x-axis, determines the threshold signal/noise ratio for a channel without storage, while the dotted lines determines that for a detector with cumulative decision-making rule: "2 of n" and "3 of n."

As can be seen from comparison of the given curves, a detector with cumulative decision-making rule is slightly inferior in noise stability to a channel with estimation; thus, for example, at $n = 10$ this advantage does not exceed approximately 1 dB.

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GEOPHYSICS, ASTRONOMY AND SPACE

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ADAPTIVE DETECTION OF SIGNALS IN A MULTIBEAM CHANNEL WITH ROUGH PREDICTION OF ITS PARAMETERS

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 122-123

[Article by A. G. Golubev]

[Text] The problem of using the results of predicting a situation (vector of parameters $\vec{\Phi} = [\Phi_1, \Phi_2, \ldots \Phi_3]$, where Φ_i is the factor of focusing an i-th beam) to improve the operating efficiency of detectors constructed by the "sliding empirical algorithm" (SEA) principle, is considered [1]. Postulation of the problem and the notations correspond to [1]. The channel is characterized by a vector of parameters $\vec{c}_j = [H_{1j}, \ldots H_{bj}; \varphi_{1j}, \ldots, \varphi_{bj}]$, where H_{ij} and φ_{ij} are random paired independent values (the amplitude and phase of the j-th signal of the detected packet in the i-th beam). The distribution density of vector c_j is a priori known

$$P_{\Phi}(\vec{c}_j) = \prod_{i=1}^{b} P(H_{ij})\,P(\varphi_{ij}) = \prod_{i=1}^{b} \frac{2^b}{(2\pi)^b} \cdot \frac{H_{ij}}{\Phi_i} \exp\left\{-\frac{H_{ij}^2}{\Phi_i}\right\}. \tag{1}$$

The result of predicting $\vec{v} = [v_1 \ldots v_b]$ of the unknown vector Φ_i is used as the vector of the parameters of a priori multidimensional exponential distribution

$$P_{v_i}(\Phi_i) = \frac{1}{v_i} \exp\left\{-\frac{\Phi_i}{v_i}\right\}; \quad P_{\vec{v}}(\vec{\Phi}) = \prod_{i=1}^{b} P_{v_i}(\Phi_i). \tag{2}$$

The decision-making procedure is constructed similar to [1] and the criterion of maximum a posteriori probability is used in calculating estimate $\tilde{\vec{\Phi}}$. A block diagram of the derived algorithm is presented in Figure 1 (the given algorithm corresponds to SEA-1 [1]).

Relations are found which permit calculation of the working characteristics of the considered algorithm during Gaussian approximation of the density of the process $\gamma(t)$. The following conclusions can be made on the basis of the given calculations.

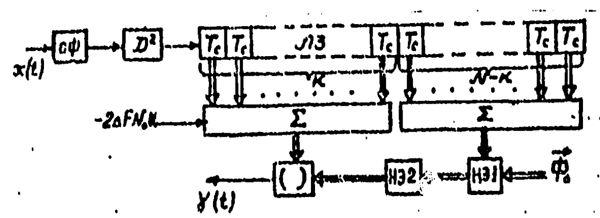

Figure 1: SF -- filter matched to one signal of the initial periodic packet; D^2 -- quadratic detector; LZ -- delay line for length NT_s with taps; T_s -- tracking period; N -- number of signals in the initial packet; N - k -- number of signals from n used as the teaching sequence; NE1 and NE2 -- nonlinear elements with amplitude characteristics for i-th reading of the estimate, respectively

$$U_{\text{вых}i} = \frac{(N-k)\vartheta_i}{2}\left[\sqrt{1+\frac{4}{(N-k)^2\vartheta_i}U_{\text{вх}i}}-1\right]-\sigma_\omega^2; \quad U_{\text{вых}i} = \frac{U_{\text{вх}i}}{U_{\text{вх}i}+\sigma_\omega^2},$$

σ_ω^2 is the noise output at the SF output; the double arrows show the vector couplings; (·) is an operation of calculating the scalar product.

Use of prediction does not lead to a reduction of the threshold signal/noise ratio. A similar result was also found in [2], where the problem, similar to our problem to some extent, is considered. In our case the result is explained by the shift of the estimate of the maximum a posteriori probability. Construction of the detector by the proposed principle yields the following advantages:

-- it permits time compression of a multibeam signal;

-- it permits identification of the channel.

The first of these effects is determined by the fact that the response of the given detector to the useful signal has a maximum only at the moment the signals traveling along separate beams coincide with their corresponding values of prediction (this combination occurs in NE1; at all remaining moments the prediction disagrees with the useful signal. When working with several values of prediction $\vec{\vartheta}_1$, $\vec{\vartheta}_2$, ..., corresponding to different situations, one may determine which situation is more probable at a given moment; thus, a prerequisite is created for solving identification problems.

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GEOPHYSICS, ASTRONOMY AND SPACE

UDC 519.282

NONPARAMETRIC ALGORITHM FOR IDENTIFICATION OF HYDROACOUSTIC SIGNALS

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 127-129

[Article by K. T. Protasov]

[Text] Solution of problems of identifying hydroacoustic situations in two stages is now proposed. Uniform linear display of the initial description (random processes, vector processes and fields) in a space of features is carried out during the first stage and the resolving rule is constructed during the second by methods of nonparametric statistics. Let us assume that a random (centered, second-order) vector field $\vec{\xi}(\vec{u}) = (\xi^1(\vec{u}), \ldots, \xi^l(\vec{u}))^t$ of vector argument $\vec{u} = (u^1, \ldots, u^v)^t$, where t is a sign of transposition, is represented in the domain of determination $D = \{\{\vec{u}\} : t_i \leq u^i \leq T_i,\ i = 1, \ldots, v\}$ by the aggregate of realizations of all classes $\vec{\xi}_1(\vec{u}), \ldots, \vec{\xi}_\lambda(\vec{u})$. Let us represent the field $\vec{\xi}(\vec{u})$ in the following manner [1]:

$$\vec{\xi}(\vec{u}) = \lim_{k\to\infty} \sum_{i=1}^{k} x^i \vec{\varphi}_i(\vec{u}), \tag{1}$$

where $\vec{\varphi}_i(\vec{u}) = (\varphi_i^1(\vec{u}), \ldots, \varphi_i^l(\vec{u}))^t$, $i = \overline{1, k}$ are the basic elements, and the random elements $\{x^t\}_k^1$ are determined by the scalar product

$$x^i = (\vec{\xi}(\vec{u}), \vec{\varphi}_i(\vec{u})) = \int_D \vec{\xi}^t(\vec{u})\vec{\varphi}_i(\vec{u})\,d\vec{u}, \tag{2}$$

and

$$(\vec{\varphi}_i(\vec{u}), \vec{\varphi}_j(\vec{u})) = \int_D \vec{\varphi}_i^t(\vec{u})\vec{\varphi}_j(\vec{u})\,d\vec{u} = \delta_{ij};\quad i, j = \overline{1, k}; \tag{3}$$

where δ_{ij} is a Kronecker symbol. The basic elements $\{\vec{\varphi}_i(\vec{u})\}_k^1$ can be found from the condition of minimum mean square criterion of approximation of the field by segment of series (1) of "k" terms, taking into account restrictions (3)

$$\varepsilon_k^2 = M\|\vec f(\vec u) - \sum_{i=1}^{k}(\vec f(\vec u), \vec\phi_i(\vec u))\vec\phi_i(\vec u)\|^2, \tag{4}$$

where M is a sign of the mean value and $\|\cdot\|$ is the norm in the sampling space. Minimization of ε_k^2 leads to the integral equation

$$\lambda\vec\phi(\vec u) = \int_D M[\vec f(\vec u)\vec f^T(\vec v)]\vec\phi(\vec v)d\vec v \tag{5}$$

(λ is a Lagrange multiplier), the approximate solution of which is found by using the algorithm for constructing the basis, adapted in the mean value [2]: let us take that one of s_j orthonormalized elements $\vec\phi_{s_j}(\vec u)$ found by the process of Gran-Shmidt orthogonalization (argument $\vec u$ is subsequently omitted), as the next basic element $\vec\phi_j(\vec u)$

$$\vec\phi_{s_j} = \frac{\vec\phi^*_{s_j}}{\|\vec\phi^*_{s_j}\|}, \quad \vec\phi^*_{s_j} = \vec f_{s_j} - \sum_{i=1}^{j-1}(\vec f_{s_j}, \vec\phi_i)\vec\phi_i, \tag{6}$$

for which

$$\lambda_j = M[(\vec f, \vec\phi_j)^2] = \max_{s_j} M[(\vec f, \vec\phi_{s_j})^2], \quad s_j = \overline{1,n}; \; j = \overline{1,k}. \tag{7}$$

The process of finding the basic elements ends in the k-th step as soon as the given accuracy $\hat\varepsilon_k^2$ of approximating the field $\vec f(\vec u)$ by linear combination "k" of basic elements $\{\vec\phi_j(\vec u)\}_1^k$ is achieved and in this case the set of random elements -- expansion coefficients $\{x^t\}_1^k$ -- is determined by formula (2). This permits one to turn from the space of signals $\vec f(\vec u)$ to the space R^k of features $\vec x = (x^1, \ldots, x^k)$.

Further assume that we know the a priori probabilities p_i and that there are teaching samples Q of classes with volume N_i, $i = \overline{1, Q}$ in R^k. As is known [3], the optimum resolving rule in the sense of a minimum mean probability of classification errors for relating vector x to the i-th ($i = \overline{1, Q}$) class has the form

$$p_i f_i(\vec x) \geq p_j f_j(\vec x), \quad j = \overline{1,Q}; \; j \neq i, \tag{8}$$

where $f_i(\vec x)$ -- the arbitrary functions of probability density are generally unknown. Let us use the following nonparametric estimate $\hat f_i(\vec x)$ for the sampling values $\{\vec x_j\}_1^{N_i}$, $\tau' = \overline{1, Q}$:

$$\hat f_i(\vec x) = \frac{1}{N_i}\cdot\frac{(c_i')^{-k}N_i^{\gamma}}{(2\pi)^{k/2}|\Sigma_i|^{1/2}}\sum_{j=1}^{N_i}\exp\{-\tfrac{1}{2}(c_i')^{-2}N_i^{2\gamma}(\vec x - \vec x_j)^T\Sigma_i^{-1}(\vec x - \vec x_j)\}, \tag{9}$$

where $1/2 > \alpha > 0$, c^{t} is a constant parameter and Σ_{t} is a covariation matrix estimated by a sample of t-th class. The empirical estimate of the mean probability of classification errors with regard to (8) and (9) then assumes the form

$$R(\vec{c}) = \sum_{i=1}^{a} \sum_{j=1}^{a} \frac{1}{N_j} \sum_{v=1}^{N_j} p_j \{ \prod_{\substack{l=1 \\ l \neq i}}^{a} E[p_i \hat{f}_i(\vec{x}_v) - p_l \hat{f}_l(\vec{x}_v)] \}, \tag{10}$$

where $E(x) = \begin{cases} 1, & z > 0 \\ 0, & z < 0 \end{cases}$, and if $j = 1$ in calculating $\hat{f}_e(\vec{x}_v)$ at point $\vec{x}_v$, the latter is eliminated from N_e sampling values, $v = \overline{1, N_e}$ (the "sliding" test). Functional (10) and at the same time the resolving rule (8) may be predetermined by optimizing (10) by the vector of unknown parameters $\vec{c} = (c^1, \ldots, c^Q)$. Assuming that the boundaries a_t, b_t of variation of components c^t of vector $\vec{c}$ are determined (for example, by the maximum plausibility method), the search for the extreme function $R(\vec{c}) \in [0, 1]$ by $\vec{c}$ will be accomplished by the Monte-Carlo method. The simplest search [4] for point $\vec{c}$ of the absolute maximum of function $\Phi(\vec{c}) = 1 - R(\vec{c})$ consists in the fact that a sequence of test points $\{\vec{c}_t\}_m^1$ with distribution $P(c') > 0$, $\forall \vec{c} \in K^a$ is given in $K^a = [a_1, b_1']x\ldots x[a_a, b_a)$. The value of the function at point $\vec{c}_{io}$ determined from condition $\Phi(\vec{c}_{io}) = \max\limits_{1 \leqslant t \leqslant m} \Phi(\vec{c}_i)$, is taken as solution of the problem. However, it is assumed that $P(\vec{c}) = \prod\limits_{t=1}^{Q} P(c^t) \equiv \equiv 1b\overline{K^a}$, but the search process can be improved if $P(\vec{c})$ is changed in each series of m experiments with regard to the values of $\Phi(\cdot)$ already found. For this purpose let us order the values of $\{\Phi(c_t)\}_m^1$ found at the test points $\{\vec{c}_t\}_m^1$, having selected m largest of them $\Phi_{(1)} \geqslant \ldots \geqslant \Phi_{(m)}$ and let us construct the density function $P(\vec{c}) = \prod\limits_{t=1}^{a} p(c^t), \vec{c} \in K^a$ in the following manner (let us carry out the construction for $P(c^t)$ at $[a_t, b_t]$, having omitted superscript t): let us introduce the parametric function

$$\psi(c) = \sum_{i=0}^{m-1} g_i f_i(c, c_i, c_{i+1}), \quad f_i(c, c_i, c_{i+1}) = \begin{cases} 1, & \text{при } c \in [c_i, c_{i+1}), \\ 0, & \text{при } c \notin [c_i, c_{i+1}); \end{cases}$$
$$c_i = \frac{c_{(i+1)} + c_{(i)}}{2}; \quad i = \overline{1, (m-1)}; \quad c_{(1)} \geqslant c_0 = a; \quad c_{(m)} \leqslant c_m = b; \tag{11}$$

$c_{(1)} \leqslant \ldots \leqslant c_{(m)}$ is the variance series of the corresponding component of the test points $\vec{c}_1, \ldots, \vec{c}_m$

$$g_i = \Phi_{(i+1)}; \quad i = \overline{1, (m-1)}; \; g_0 = \Phi_{(1)}; \; g_m = \Phi_{(m)}.$$

The desired density function has the form

$$p(\vec{c}) = \prod_{t=1}^{a} \frac{1}{S_t} \psi(c^t), \tag{12}$$

where the normalizing multiplier $S_{\dot{t}} = \int_{a_t}^{b_t} \psi(t)dt$ and $p(\vec{c})$ has "increased" values of density in the range of those values of $\vec{c} \in \overline{R}^{n}$ at which the highest values of $\phi(\vec{c})$ occurred in the previous series of M tests. The next series of M test points $\{\vec{c}_{\tau}\}_{m}^{1}$ has distribution (12). Thus, the search process is divided into a series of tests by M with rearrangement of $p(\vec{c})$ on each series with regard to the best results of the previous series.

The quality of the tracked extreme value will be characterized by two moments of (primary) order statistics of extremes. Let us use the distribution of order statistics of extremes [5]; the sample mean value $\bar{\mu}$ and the sample dispersion $\bar{D}$ can be found as estimates of linear functionals:

$$\bar{\mu} = \sum_{j=1}^{M} \left(\frac{j}{M}\right)^{M-1} \cdot \phi_j, \quad \bar{D} = \sum_{j=1}^{M} \left(\frac{j}{M}\right)^{M-1} \cdot (\phi_j - \bar{\mu})^2, \tag{13}$$

where $\{\Phi_j\}_{m}^{1}$ are the values of function $\Phi(\vec{c})$ found in a series of M tests.

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GEOPHYSICS, ASTRONOMY AND SPACE

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OPTIMUM NONLINEAR EXTRAPOLATION FOR RANDOM FIELDS

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 130-131

[Article by B. S. Shtatland]

[Text] The results of solving problems of filtration, interpolation and extrapolation of random fields found by some dynamic equations are presented in [1]. In this case the extrapolation problem is understood as construction of an optimum estimate of some unobserved field at moment of time $t + \tau$, where τ = const, by observation on the interval [0, t] of another field statistically related to the unobserved field. The stochastic differential relation satisfied by the optimum estimate of the unobserved field by the minimum mean square error at some fixed moment of time T, $t < T$, is presented in this communication.

We will retain the same terminology as in [1]. The results of [2] are used extensively in proof of the theorem presented below and the theorem itself is generalization of theorem 8.5 from [3] for random processes with values of H and, consequently, for random fields.

Let random processes η and ξ with values of H or some expansion of it H_- (see [2]) and the nondecreasing flow of σ-algebras $F_u \subset F$, with respect to which all the considered processes are measured, be given on a fixed probability space (Ω, F, P). The process $\eta = (\eta_u, F_u)$ has the form:

$$\eta_u = \eta_0 + \int_0^u B_s \, ds + x_u , \tag{1}$$

where $x = (x_u, F_u)$ is the quadratically integrated martingale with values of H or H_-, $B = (B_u, F_u)$ is some random process with values of H such that

$$\int_0^T \|B_s\| ds < \infty, \qquad (P\text{-п.н.}) \tag{2}$$

We shall call process η the unobserved process (it can be interpreted as a signal).

The process $\xi = (\xi_u, F_u)$ having the following form is observed:

$$\xi_u = \xi_0 + \int_0^u A_s \, ds + w_u \, , \tag{3}$$

where $w = (w_u, F_u)$ is a Wiener process with values of H_-, $A = (A_u, F_u)$ is a random process with values of H which satisfies the condition

$$\int_0^T \|A_s\| \, ds < \infty \qquad (\text{Р-п.н.}) \tag{4}$$

and such that all the information about the unobserved process, accessible to the observer, is contained in it. Specifically, A_u may simply coincide with η_u. Also assume that

$$\int_0^T E\|B_s\|^2 ds < \infty \quad \text{и} \quad \int_0^T E\|A_s\|^2 ds < \infty . \tag{5}$$

The problem consists of constructing the optimum estimate of the unobserved process η at moment of time T, $t < T$ by observation of process ξ on time interval $[0, t]$. It is known that this estimate is the arbitrary mean value $E[\eta_T F_t^\xi]$, where $F_t^\xi = \sigma\{\xi_s, 0 \leqslant s \leqslant t\}$. The following result is valid.

Theorem. If conditions (1)-(5) enumerated above are fulfilled, $E[\eta_T F_t^\xi]$ satisfies the following stochastic relation:

$$E[\eta_T | \mathcal{F}_t^\xi] = E[\eta_T | \mathcal{F}_0^\xi] + \int_0^t \Big\{ E[D_t | \mathcal{F}_t^\xi] + E\big[E[\eta_T | \mathcal{F}_t] \otimes \big(A_t - E[A_t | \mathcal{F}_t^\xi]\big) | \mathcal{F}_t^\xi\big]\Big\} d\nu_t , \tag{6}$$

where $D_t = (d \langle \tilde{x}, w \rangle_t)/dt$ and $\tilde{x} = (\tilde{x}_t, F_t)$ is the quadratically integrated martingale with $\tilde{x}_t = E[\eta_t F_t]$ and $\langle \cdot , \cdot \rangle$ is a sign of the mutual structural function of two martingales, $\otimes$ is a sign of the tensor product and

$$d\nu_t = d\xi_t - E[A_t | \mathcal{F}_t^\xi] \, dt.$$

We note that $\nu = (\nu_t, F_t^\xi)$ is a Wiener process with values of H_-.

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GEOPHYSICS, ASTRONOMY AND SPACE

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THE UNBIASED RULE OF CLASSIFYING THE STATE OF A SONAR OBJECT WITH PARAMETRIC A PRIORI AMBIGUITY

Novisibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 132-135

[Article by Yu. Ye. Sidorov]

[Text] One of the problems with parametric a priori ambiguity is that of classifying the state of the object (fixed or moving) which may also be regarded as a problem of detecting a moving object. It is solved on the basis of observing the aggregate of marks from the object which are random values with known distribution law, but with unknown parameters of this law. Solution of this problem is also important for the problem of determining a fixed target on a noisy background [1].

One of the variants of solving the problem of synthesizing the rule (and of the corresponding device) of detecting a moving object (or the rule of classifying the state of an object being located: fixed or moving) under conditions of parametric a priori ambiguity is considered in the given paper. The following requisites are used to solve this problem.

1. There are two combinations of independent marks with coordinates $\{x_{1i}; y_{1i}\}$ and $\{x_{2i}; y_{2i}\}$, $i = 1, 2, \ldots, n$, which may be regarded as coordinates of the instantaneous position of the object, found during two adjacent periods of repetition of the sounding signal by using a system of separated data sensors.

2. Distribution of each of the independent combinations $\{x_{1i}\}$, $\{y_{1i}\}$, $\{x_{2i}\}$ and $\{y_{2i}\}$ is assumed normal [4, 5, 6] with unknown mean values (which characterize the location of the object) and dispersion.

3. An informative feature of the motion of an object is the increment of mean values of observed marks during the adjacent period of signal transmission by value Δ. In the opposite case the solution of the fixed nature of the object being located is taken. The increment Δ due to the unknown speed of motion of the object is assumed unknown and may be different for each of the coordinates. Since the combinations $\{x_{1i}; x_{2i}\}$ and $\{y_{1i}; y_{2i}\}$ are

assumed independent with solution of secondary processing problems [2, 3, 4], it is sufficient to find the detection algorithm for one of them, for example, for $\{x_{1i}; x_{2i}\}$ since its structure will be similar for the second one.

Based on the adopted requisites, the problem of detection (classification) may be formulated thusly: one must determine whether the object being located is fixed or moving by two samples $X_1 = x_{1i} \in N(\xi_i, \sigma^2)$ and $X_2 = x_{2i} \in N(\xi_i + \Delta, \sigma^2)$ (parameters ξ_i, σ^2 and Δ are a priori known). This problem is equivalent to that of checking hypothesis H: $\Delta = 0$ (the object is fixed) with respect to the alternative K: $\Delta > 0$ (the object is moving).

The optimum rule for checking hypotheses H and K should satisfy the following practically important requirements.

1. The test $\Phi(X_1, X_2)$ corresponding to the decision rule, should not depend on a priori unknown distribution parameters of observed marks.

2. The probability α of a false signal should be constant at any values of unknown parameters.

3. The probability β of a correct decision should be maximum.

The rule of classifying the state of the object being located (fixed or moving), which satisfies the formulated requirements, should belong to the class of unbiased (similar in the range of hypothesis H) uniformly more powerful (RNM) rules for checking two complex statistical hypotheses [5, 6].

The structure of the postulated problem becomes clear if the following orthogonal transformation is carried out [5]:

$$x'_{1i} = (x_{2i} - x_{1i})\sqrt{2}, \qquad x'_{2i} = (x_{1i} + x_{2i})/\sqrt{2}.$$

The rule for checking H and K may then be determined in terms of variables $w_i = x_{2i} - x_{1i}$. This is the unbiased RNM (RNMN) with critical range of type:

$$V = \frac{\sqrt{n}\,\bar{w}}{\sqrt{\frac{1}{n-1}\sum_{i=1}^{n}(w_i - \bar{w})^2}} > C, \tag{1}$$

where $\bar{w}$ is the mean value of w_i, i = 1, 2, ..., n. Threshold C is determined by the given level of significance, i.e., in the given case by the given probability α of a false decision in favor of the hypothesis from the condition:

$$\alpha = \int_{C}^{\infty} t_{n-1}(v)\,dv, \tag{2}$$

where $t_{n-1}(v)$ is the statistical distribution V with hypothesis A, which is the central Student t-distribution with (n - 1)-degree of freedom.

The block diagram of the classification device is determined from the following expression, equivalent to [1]:

$$C_1\left[\sum_{i=1}^{n}(x_{2i}-x_{1i})\right]^2_{c^2} - C_2\sum_{i=1}^{n}(x_{2i}-x_{1i})^2 + C_3\left[\sum_{i=1}^{n}(x_{2i}-x_{1i})\right]^2 > 0,$$

where $C_1 = 1/n$, $C_2 = c^2/n-1$ and $C_3 = c^2/n(n-1)$ are also shown in Figure 1. The following is denoted here: UPO -- primary processing device from whose output marks from the object are taken; BVR -- block for calculating the differences $x_{2i} - x_{1i}$; $\sum$ -- adder; UVK -- device for raising to the power of 2; [translator's note: one letter missing from text] -- block for multiplying by the constant C_i, $i = 1, 2, 3$; "-" -- subtraction device; "+" -- adding device. Realization of this scheme is possible in the digital variant.

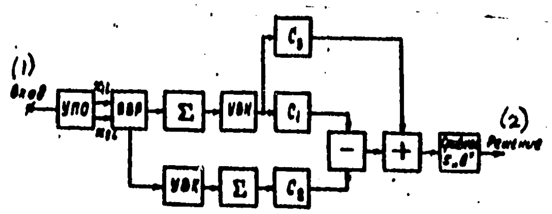

Figure 1. Block Diagram of Device for Classifying the State of an Object

KEY:

1. Input
2. Decision

The effectiveness of rule (1) and (2) is determined by the power function of the RNMN criterion, equal to

$$\beta(\delta) = \int_{c}^{\infty} t_{n-1}(v, \delta)\,dv, \tag{3}$$

where $t_{n-1}(v, \delta)$ is statistical distribution V with alternative K, which is the noncentral Student t-distribution with (n - 1)-degree of freedom and noncentral parameter $\delta = \sqrt{n}\,\Delta/\sqrt{2}\,\sigma$. Probability (3) may be calculated by using the tables of [7], for example.

It should be noted that the sign of inequality in (1) changes to the opposite when checking hypothesis H with respect to alternative K: $\Delta < 0$.

Thus, the RNMN rule for classifying the state of an object, fixed or moving, was found (rule for detecting a moving object), having very desirable properties for automation of processing, namely: a) the rule does not depend on a priori unknown parameters $\xi_1, \ldots, \xi_n, \sigma^2$ and Δ; b) it guarantees the constant probability α of a false decision and in this case threshold C does not depend on $\xi_1, \ldots, \xi_n, \sigma^2$ and Δ and does not need readjustment during operation of the device to the actual values of these values; and c) they have maximum probability of correct decision in the class of all unbiased rules. The noted properties permit one to use this rule in automatic digital sonar data processing systems under conditions of a priori ambiguity.

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GEOPHYSICS, ASTRONOMY AND SPACE

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AN ADAPTIVE DETECTOR WITH DIGITAL SPACE-TIME PROCESSING OF NOISE SIGNALS

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 136-137

[Article by Yu. P. Podgayskiy and A. M. Yakubovskiy]

[Text] Detectors with quantification of the continuous input sample after quantification of it at the outputs of a K-channel antenna by using an analog-digital converter (ATsP) system has found wide application in signal processing. The additive noise at ATsP outputs, uncorrelated between each other, may be partially suppressed in an adaptive detector in the same manner as in a nonadaptive detector. Additional suppression, based on the use of the spectral signal and noise differences, can be accomplished in a post-adder filter and eliminated from processing in the multichannel part of the detector by introduction of the so-called vector of limitations G into the adaptive system.

Let the spectral characteristics of a signal be a priori unknown in the given direction of observation. Spatial processing is optimized on the basis of maximizing the power increment of the observed output process due to the appearance of a signal normalized to noise output $(W^TRW - W^TR_NW)/W^TR_NW$ or $[W^T(R - R_N)W]/W^TR_NW$ provided that $C^TW - G = 0$, where R and R_N are covariant matrices of the signal and noise mixture and of noise only, respectively. All the processes are Gaussian with zero mean value uncorrelated between each other. W is a weight vector of a space-time processing system for processes at I moments of time simultaneously. Vector W with dimension KI may be found by the Lagrange method, on the assumption that the signal at the output is much less than noise, i.e., $tr(R - R_N) \ll tr\bar{R}_N$ and $W^TR_NW = WRW$. The optimum vector is found in the form:

$$W = W_0 = R_N^{-1}(R-R_N)C[C^TR_N^{-1}(R-R_N)C]^{-1}G,$$
$$C = diag(C_1, \ldots, C_j, \ldots, C_J), \qquad C_j = [C_{j1}, \ldots, C_{jK}]^T.$$

Let us find the iterative procedure for optimization of a detector during its functioning. Let us introduce the vector of limitations to the control algorithm for space-time signal processing. Let us use the fastest start and Lagrange methods. Using the estimate of the unknown covariant matrices R and R_N in the form of the scalar product of the input vectors of the space-time sample of random processes of the system oriented in the direction of signal observation and in the direction where the signal is known to be absent, respectively, at each iteration, we find the stochastic control algorithm:

$$W(n+1) = P\left\{W(n) - \mu \frac{1}{y_1^2}\left[yX(n) - y_1 X(n)\right]\right\} + C(C^T C)^{-1} G$$
$$W(0) = (CC^T)^{-1} C G, \qquad P = I - C(C^T C)^{-1} C^T, \tag{1}$$

where $y = W^T(n)X(n)$ is the estimate of the signal in the direction of its observation and $y_1 = W^T(n)X_1(n)$ is the estimate of the process during observation in the direction in which the signal is known to be absent (outside the direction of observation).

The weight vector of limitations may be found by using the iterative procedure, for example, presented in [1] for optimization of a post-adder filter during its functioning. In this case the filter should accomplish weight processing of the processes at the appropriate moments of time I simultaneously. The vector of limitations is numerically equal to the optimum weight vector of the weight post-adder filter at whose input there is time sampling of the output process of the space-time processing system oriented in the direction of signal observation.

The output process is random with zero mean value. The weight vector of the filter is determined on the basis of the minimum mean square error in the estimate of the signal in the adder. The optimum weight vector by the given criterion is found in the form:

$$G = V = V_0 = \Phi_Y^{-1} \Phi_{Yd} .$$

Here Φ_Y is the covariant matrix of the I-dimensional vector Y of the time sample in the adder and Φ_{Yd} is the vector of the covariant vector Y with the required signal at the filter output with the anticipated value of dispersion σ_d^2.

The vector of limitations in algorithm (1) is constant (or slowly variable), which can be calculated beforehand (or at long intervals). The operation of multiplication by matrix P reduces to several additions since matrix C has simple form.

The algorithm requires a small number of multiplication and memory operations (proportional to the number of weight coefficients KI). It can be used in large arrays.

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SIMULATION MACHINE EXPERIMENTS IN STATISTICAL HYDROACOUSTICS: MODELS, ALGORITHMS AND MEASUREMENTS

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 138-153

[Article by V. V. Ol'shevskiy]

[Text] 1. Preliminary comments. Modeling of various types of phenomena and systems using computers (EVM) has been used extensively during the past few years in many fields of sciencd and technology (see, for example, [1-5]). Especially strong progress in the field of modeling has been observed during the past few years with regard to the increase of the information capacity of computers: actually, such problems are now being formulated and practically solved which were regarded as hopelessly cumbersome, unconstructive and requiring considerable simplification 10-15 years ago.

One of the natural manifestations of the general trends of ever broader use of computers in investigations should be regarded the occurrence of a new field -- simulation machine experiments [5]. In statistical hydroacoustics, simulation machine experiments are understood as numerical methods of conducting statistical measuring experiments on computers with mathematical models of the hydroacoustic objects of investigation (phenomena and information systems); these objects are observed over prolonged time intervals and with different combination of hydrophysical conditions when conducting simulation machine experiments.

The following main steps are considered when conducting simulation machine experiments [5, 10]:

-- determination of the purpose of the investigations;

-- selecting the model of the investigated object;

-- compiling the computer programs;

-- planning the simulation experiment;

-- processing the results of the experiment and interpretation of them.

We shall consider below some characteristic features of these types of simulation machine experiments with respect to statistical hydroacoustics problems.

2. Mathematical models. Selection of the initial mathematical model is the most important item in conducting simulation machine experiments in statistical hydroacoustics. The success of modeling as a whole depends in the final analysis on the extent to which the adopted model is adequate to the real objects of investigation (close or similar to them) and the extent to which it is constructive (simplicity, profitable and realizable).

We shall further discuss some methodological problems related to the characteristics of selecting mathematical models of random fields required to conduct simulation machine experiments in statistical hydroacoustics.*

According to established concepts [6-10], the mathematical model of a hydroacoustic field is the representation of it on the basis of the given hydrophysical characteristics of the water medium and its boundaries which permits one to determine the probability characteristics of the field essential in the problem being solved. The mathematical model is an idealized image of real hydroacoustic phenomena, description of which is given on the basis of theoretical-probable concepts and methods. Thus, if $X(t, \vec{r})$ is a random hydroacoustic field and $\theta(e/m)$ is its probable characteristic corresponding to some model m, then to determine the model of the field means to give the following concepts:

$$X(t,\vec{r}) = \mu_{\xi}\{\xi_i(t,\vec{r})\}, \qquad (1)$$

$$\theta(\vec{\ell}/m) = \mu_{\pi}\{\pi_i(\vec{\ell})\}, \qquad (2)$$

where $\xi_i(t, \vec{r})$ are the initial hydrophysical fields of the water medium and its boundaries; $\pi_i(l)$ are the probability characteristics of these fields which are assumed given (input data); $i = \overline{1, R}$; and μ_{ξ} and μ_{π} are operators of formation, respectively.

To be given the model m means to determine it in the following manner:

$$m: \{\theta_j(\vec{\ell}/m)\} = \{\theta(\vec{\ell}, \vec{a}_j/m), P(j, \vec{a}_j)\}; \ \vec{a}_j \in A_m; \ j = \overline{1, N_m}, \qquad (3)$$

where $\theta_j(\vec{l}/m)$ are the probability characteristics corresponding to the model m; a_j are parameters which describe the hydrophysical situation (refraction, attenuation, scattering and so on); A_m and N_m are the space and number of parameters $\vec{a}_j$; $P(j, a_j)$ is the joint probability distribution of numbers j of

*Special cases of random fields are of course random processes and values.

characteristics $\theta_j(l/m)$ and of their parameters a_j. It follows from this interpretation that each model of the hydroacoustic field corresponds to a parametric set of probability characteristics.

The set M of all models m, which investigators may have at their disposal,

$$m \in M \tag{4}$$

forms the archive of models (they also sometimes say the thesaurus, library or file of models, which is equivalent in content). The archive of models M can be divided into two subsets M_1 and M_2, so that

$$M = M_1 \cup M_2; \tag{5}$$

$$m_1 \in M_1; \quad m_2 \in M_2, \tag{6}$$

where M_1 is the set of ideal (true) models m_1 and M_2 is a set of working models m_2.

Ideal models m_1 correspond to complete probability description of the investigated hydroacoustic field, i.e., to being given infinitely dimensional probability densities or characteristic functions. These models should accurately and ideally describe the properties of the real hydroacoustic objects. Therefore, it is natural that they are never unknown to the investigator, but we assume that they objectively exist. The working models m_2 describe the hydroacoustic objects partially, they are incomplete, but in this case they describe these objects rather well (from the practical viewpoint) and, which is especially important, are accessible for realization in modeling, experiments and measurements.

The next problem consists in correctly approaching selection of the model of investigated hydroacoustic objects by using clear quantitative criteria. For this purpose let us introduce the following two characteristics:

-- the measure of adequacy on the part of the working model m_2 with respect to the ideal model m_1:

$$\rho_\theta(m_1, m_2) = \rho[\theta(\vec{l}/m_1), \theta(\vec{l}/m_2)];$$

-- the measure of expenditures in realizing the working model m_2:

$$C_\theta(m_2) = C[\theta(\vec{l}/m_2)]. \tag{8}$$

In this case the value of $\rho_\theta(m_1, m_a)$ is the distance of probability characteristics $\theta(\vec{l}/m)$ in functional space which correspond to the ideal m_1 and working m_2 models, while $C_\theta(m_2)$ is a functional which characterizes the degree of complexity of working model m_2 when solving the postulated problem at the level of probability description $\theta(\vec{l}/m)$.

As already noted, the ideal (true) model m_1 is never unknown to the investigator and we can only postulate its objective existence. Therefore, it is natural in further consideration to replace m_1 by another model m_2^* which would satisfy the condition:

$$\rho_0(m_1,m_2^*)\leq\Delta\rho^*, \tag{9}$$

$$m_2^*\in M_2^*\subset M_2, \tag{10}$$

and $\Delta\rho^*$ is a sufficiently small value. This model m_2^* was called the metamodel [10]. The only method of determining the value of $\Delta\rho^*$ and consequently of metamodel m_2^* is apparently direct full-scale experiments to establish the conformity of the predicted probability characteristics $\theta(\vec{l}/m)$ and their statistical estimates $\hat{\theta}(\vec{l})$ obtained experimentally. In this case the value of $\Delta\rho^*$ is determined, for example, as the mean square difference

$$\langle[\hat{\theta}(\vec{l})-\theta(\vec{l}/m_2^*)]^2\rangle\leq\Delta\rho^*.$$

Let us present two examples to explain the content of the concept of the metamodel of the investigated hydroacoustic object. The first example concerns modern measurement theory [11] where the metamodel of the measured object is understood as that working model (accessible to measurement) which is as close to this object (adequate to it) as is dictated by the needs for solving the measuring problems of given class. The second example is related to mathematical logic [12] whose objects of investigation are propositions (inferences) on which operations similar to operations on numbers are carried out; mathematical logic is sometimes called metamathematical logic.

Thus, the set of working models m_2 is formed by the set of metamodels M_2^* and by the set

$$M_2^{**}=M_2\setminus M_2^* \tag{11}$$

which realized the working models which are simpler than the metamodels and are accessible for modeling and practical use.

Two types of problems of selecting the optimum working model of a hydroacoustic object to model it can be formulated in view of the measure of adequacy (7), measure of expenditures (8) and the metamodel introduced above.

The first type of problems are:

-- limitation of expenditures on modeling is introduced, i.e., it is assumed that

$$C_2(m_2^c) \leq \Delta C; \tag{12}$$

and in this case the set M_2^c of permissible working models is determined from (12)

$$m_2^c \in M_2^c \subset M_2^{**}; \tag{13}$$

the optimum working model opt m_2 is selected on the basis of the following condition:

$$\text{opt}\, m_2 = \arg \inf_{m_2 \in M_2^c} \rho_2 (m_2^{*}, m_2). \tag{14}$$

In this postulation the optimum model with which simulation machine experiments will be conducted are assumed more adequate than the metamodel with regard to limitations on expenditures during modeling.

The second type of problems is:

-- limitation on the permissible measure of adequacy is introduced, i.e., it is assumed that

$$\rho_2(m_2^{*}, m_2^{\rho}) \leq \Delta \rho; \tag{15}$$

in this case the set M_2 of permissible working models is determined from (15)

$$m_2^{\rho} \in M_2^{\rho} \subset M_2^{**}; \tag{16}$$

the optimum working model opt m_2 is selected on the basis of the following condition:

$$\text{opt}\, m_2 = \arg \inf_{m_2 \in M_2^{\rho}} C_2(m_2). \tag{17}$$

In this postulation the optimum model with which the simulation machine experiments will be conducted is assumed as corresponding to the least expenditures for modeling with regard to the permissible measure of its non-adequacy to the metamodel.

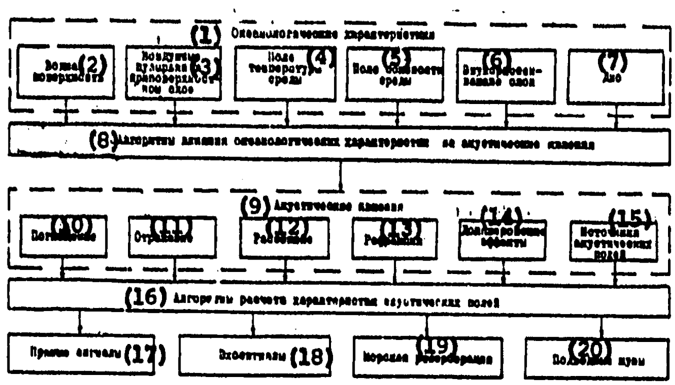

Figure 1. Diagram of Formation of Acoustic Model of the Ocean

KEY:

1. Oceanological characteristics
2. Water surface
3. Air bubbles in surface layer
4. Temperature field
5. Salinity field
6. Sound-scattering layer
7. Bottom
8. Algorithms of the effect of oceanological characteristics on acoustic phenomena
9. Acoustic phenomena
10. Absorption
11. Reflection
12. Scattering
13. Refraction
14. Doppler effects
15. Sources of acoustic fields
16. Algorithms for calculating the characteristics of acoustic fields
17. Direct signals
18. Echo-signals
19. Sea reverberation
20. Underwater noise

In light of the discussed concepts, let us return to the archive of models M. According to (4)-(6) and (10)-(11),

$$M = M_1 \cup (M_2^* \cup M_2^{**}), \tag{18}$$

$$m_1 \in M_1,\ m_2 \in M_2,\ m_2^* \in M_2^*,\ m_2^{**} \in M_2^{**}. \tag{19}$$

The nodal and central aspect in the archive of models is the set M_2^* of metamodels m_2^*. This set is expanded with deepening of our knowledge due to the set M_1 of ideal models m_1 (supplementation of theoretical and experimental data on acoustic phenomena in the ocean); it also generates an ever broader set M_2^{**} of realizable working models M_2^{**} (solution of problems of selecting the optimum working models opt $m_2 \in M_2^{**}$).

3. Some aspects of working out the mathematical acoustic model of the ocean. Let us first note the main methodical features of working out the acoustic model of the ocean and let us consider the diagram (Figure 1) of formation of this model on the basis of oceanological characteristics.

a) The mathematical acoustic model of the ocean should be a probability model. There are the following bases for this statement (it would seem that a categorical statement is unnecessary). First, the presence of clearly marked dynamic hydrophysical phenomena in the ocean essentially does not permit description of them within the framework of the determined equations; it is also practically impossible to count on determined dynamic checking of these phenomena when conducting full-scale experimental investigations. Second, the overwhelming majority of problems in statistical hydroacoustics are informative in nature, for solution of which only probability models of input effects (signals and noise) are suitable. Third, any mathematical model of the investigated object is worked out on the basis of observations or measurements of one or another of its characteristics; the results of similar investigations are essentially statistical in nature with respect to working out the acoustic model of the ocean and their theoretical analog is the probability model.

b) The initial models are the nodal probability acoustic models of the ocean. One may now assume as an established fact (see, for example, [6-10, 13-17]) that there are several stochastic mechanisms in the ocean which affect formation of acoustic fields in it. These mechanisms are distinguished by scales (in time and space) of their own effect and may be classified in the following manner.

The following are typical stochastic time mechanisms:

-- rapid fluctuations of the hydrophysical characteristics of the water medium and of its boundaries, related mainly to the dynamics of the water surface and turbulent phenomena in the medium (time scales -- from fractions of a second to tens of seconds);

-- daily variations related to variation of the thermodynamic conditions of the atmosphere and ocean (time scales -- up to several hours);

-- seasonal variations related to variation of weather conditions and change of the seasons (time scales -- months).

The following are typical spatial stochastic mechanisms:

-- small-scale variations of the hydrophysical characteristics of the medium and of its boundaries related mainly to the spatial structure of the water surface and bottom, the microtemperature inhomogeneity of the medium, the presence of inhomogeneous inclusions -- air bubbles, biological organisms and so on (spatial scales -- from millimeters to tens of meters);

-- medium-scale inhomogeneities related to the difference of climatic conditions and currents (spatial scales -- tens and hundreds of kilometers);

-- inhomogeneities of oceanic scales (thousands of kilometers).

Of course, this classification of the scales of stochastic hydrophysical mechanisms in the ocean is very rough and approximate, but it still permits one to discuss some methodological problems of constructing acoustic models of the ocean.

Based on the foregoing, any probability acoustic model obtained with respect to a specific moment of time and geographic coordinates should be regarded as a conditional model. Further expansion of the models should take into account the probability distribution of the classes of conditions and their characteristics. These types of conditional models are the initial and elementary probability models and should comprise the basis of the archive of models.

c) The archive of acoustic models of the ocean may be constructed at each step of investigations. The process of intensive investigations of the acoustic characteristics of the ocean has continued for not less than 30 years, i.e., the first substantial investigations began to appear (see, for example, [13]) at the end of the 1940's. Even at that time the main physical phenomena were determined which essentially characterize the properties of the acoustic fields in the water medium: the refraction of acoustic waves, absorption, scattering on the inhomogeneities of the medium and of its boundaries, and also a number of phenomena related to mechanisms of noise formation and reflection of acoustic waves by different bodies. There was essentially a specific model of the ocean even at that time and it was used extensively, for example, in solving a broad class of measuring and systems problems (see [18, 19] and others).

Of course, from today's concepts, the models developed in the 1950's and even in the 1960's were limited, simplified and very inadequate to real phenomena. But investigators will look just as skeptically within 10 years on those acoustic models of the ocean which are now being discussed in the middle of the 1970's.

Thus, according to the specific goals of using the models and on the basis of the essential level of investigations, the archive of acoustic models of the ocean is constructed at each step.

d) The acoustic model of the ocean should permit one to predict real acoustic phenomena with controlled accuracy. The problem of predicting the phenomena of the real world from given models of these phenomena is closely related to development of methods of finding empirical principles and has been widely discussed in the literature of the past few years (see, for example, [20]). Here we shall encounter a number of problems of the correct approach to formation of the archive of models, selection of the quantitative criteria for comparison of models with each other and for comparison of the models with experimental data. It will apparently be no exaggeration to assume that the enormous aggregate of experimental data on the acoustic characteristics of the ocean has not yet been classified from unified positions of modern methods of finding the principles, pattern recognition and checking and amplifying the empirical hypotheses. This viewpoint should of course not be regarded as unnecessarily pessimistic since the experience accumulated during the past 30 years is so diverse that the time is fully ripe to change from the previously developed energy models to probability models in the methodical sense.

In light of the discussed aspects related to development of mathematical acoustic models of the ocean, let us point out that simulation machine experiments are one of the powerful means of creating these models, they permit correct study of the problems of the adequacy of the models to real objects of investigation and, finally, they help to organize prediction of the acoustic characteristics of the ocean in the complex combination of hydrophysical factors.

4. An example of simulation machine experiments: modeling of sea reverberation. Sea reverberation was modeled on the BESM-6 computer according to the method described in [10, 21].

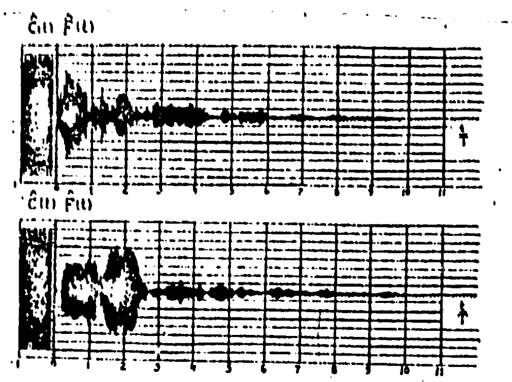

Figure 2. Sample Realizations of Reverberation. The emitted signal has square-wave envelope and sine-wave filling; the layer is scattered and the scatterers are fixed

The simulation experiments were reduced to generation of sample realizations $\hat{F}_k(t)$ of reverberation, its quadrature component $\hat{F}_{ck}(t)$ and $\hat{F}_{sk}(t)$, the envelope $\hat{E}_k(t)$ and the current phase $\psi_k(t)$. The discrete model of reverberation was taken as the initial model [6, 8]:

$$F(t)=\sum_{i=0}^{N(t)} a_i f(t_i) c(t-t_i, \bar{\varepsilon}_i), \qquad (20)$$

where N(t) is the random number of elementary scattered signals which form reverberation F(t) at moment of time t; a_i is the random amplitude of the i-th signal; t_i is the random moment of arrival of the i-th signal; $f(t_i)$ is a function which characterizes the decrease of signals with distance (in time);

$\vec{\varepsilon}_i$ are random parameters which change the shape of the scattered signals compared to the emitted signal $c(t)$. The quadrature components $F_c(t)$ and $F_s(t)$ of reverberation and also of its envelope $E(t)$ and the current phase $\psi(t)$ were found from ordinary concepts:

$$F(t)=F_c(t)\cos\omega_0 t - F_s(t)\sin\omega_0 t, \tag{21}$$

$$F_c(t)=E(t)\cos\psi(t),\ F_s(t)=E(t)\sin\psi(t), \tag{22}$$

$$E(t)=[F_c^2(t)+F_s^2(t)]^{1/2};\ \psi(t)=\mathrm{arctg}\frac{F_s(t)}{F_c(t)}, \tag{23}$$

where ω_0 is the central frequency of the emitted signal spectrum.

The digital machine model of reverberation was realized by algorithm (20) and $N(t)$ was a transient Poisson flow. It was possible to model the following variants of the occurrence of reverberation:

-- representation of different functions of the mean number $\langle N(t)\rangle$ of scattered signals which form reverberation and of different functions $f(t)$ which characterize variation of the mean signal levels during their propagation;

-- representation of various types of emitted signals $C(t)$: with sine-wave filling and different envelopes, frequency-modulated and noisy;

-- representation of various types of distributions $W(a)$ of the amplitudes a_i of elementary scattered signals;

-- representation of various types of distributions $W(V)$ of the velocities of the scatterers $\vec{\varepsilon}_i = V_i$ (in this case the Doppler effect is interpreted as variation of the time scale of elementary scattered signals rather than only as shifting of their central frequency spectrum).

Examples of sample realizations of reverberation $\hat{F}(t)$, corresponding to different input data of simulation machine experiments, are presented in Figures 2 and 3. An example of sample realizations of reverberation $\hat{F}(t)$, its quadrature components $\hat{F}_c(t)$ and $\hat{F}_s(t)$, envelope $E(t)$ and current phase $\hat{\psi}(t)$ is given in Figure 4. This case corresponds to essentially transient reverberation when the average number of scattered signals $\langle N(t)\rangle$ (see Figure 4, f) has jumps. It is interesting to note that reverberation has an appreciable coherent component, manifestation of which may be noted in the characteristic quadrature components, envelope and phase [6, 8, 9], on time intervals where the gradients $\langle N(t)\rangle$ are high.

Simulation machine experiments showed [10, 21] that all the investigated probability characteristics of reverberation, of its quadrature components, envelope and current phase, calculated on the basis of the discrete Poisson

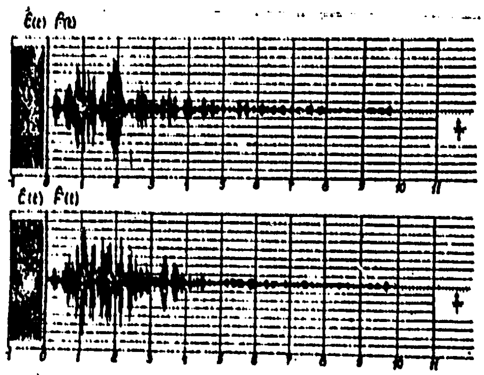

Figure 3. Sample Realizations of Reverberation. The emitted signal has square-wave envelope and frequency modulation by linear law: the layer is scattered and the scatterers are fixed

model of reverberation [6, 8], agree well with statistical estimates obtained for the modeled processes.

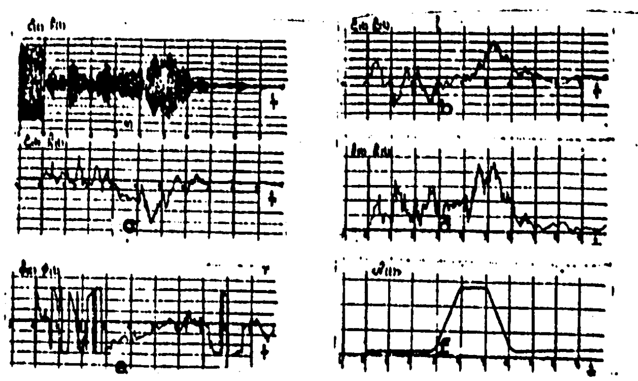

Figure 4. Sample Realizations: a -- reverberation; b and c -- quadrature components; d -- envelope; e -- current phase; f -- average number of scattered signals

5. Characteristics of statistical measurements. When conducting full-scale and machine experiments in statistical hydroacoustics, a typical situation is that when the measuring experiments generate an aggregate of sample realizations $\{\hat{X}_k(t)\}$, $k = \overline{1, N}$, each of which $\hat{X}_k(t)$ is related in the general case to its own probability model m_k. These experiments are regarded as dynamic [22] in the sense that the statistical homogeneity of the course of the investigated phenomenon is disrupted upon transition from one sample realization to another. Let us determine the one-dimensional probability characteristic $\theta(1/m_p)$ which describes the properties of sample realization $\hat{X}_p(t)$ in the following manner:

$$\theta(l/m_p) = \langle \vartheta[X_p(t), l/m_p] \rangle = \langle \vartheta[X_p(t), l] \rangle, \tag{24}$$

where v is an operator for formation of the probability characteristic of type θ and 1 is an argument of this characteristic.

Let us determine the statistical estimate $\hat{\theta}(1)$ of characteristic (24) in the form

$$\hat{\theta}(l) = \frac{1}{N}\sum_{k=1}^{N} \vartheta[\hat{X}_k(t), l], \tag{25}$$

i.e., let us average by N sample realizations; moreover, the p-th realization corresponds to the true model m_p.

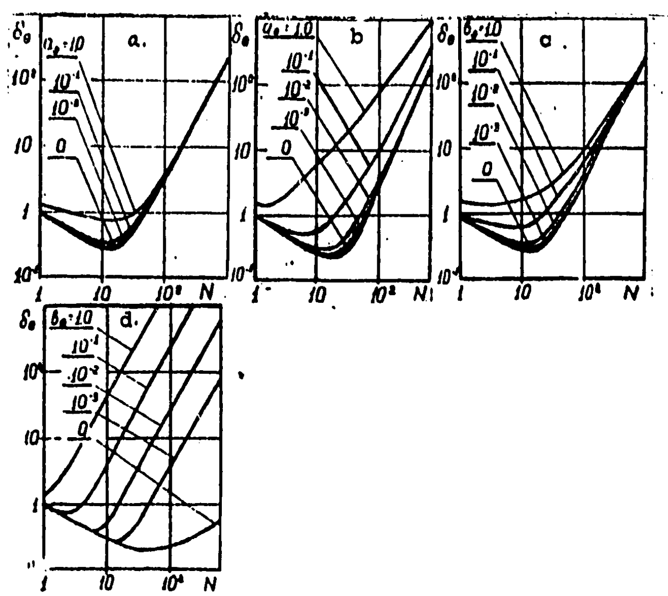

Figure 5. Dependence of Relative Total Statistical Measurement Error on Number of Sample Realizations: a -- $B_v = a_\theta = B_\theta = 10^{-3}$; b -- $a_v = B_v = b_\theta = 10^{-3}$; c -- $a_v = a_\theta = b_v = 10^{-3}$; d -- $a_v = b_v = a_\theta = 10^{-3}$

The following problem of optimizing the procedure of statistical measurements occurs under conditions of dynamic hydroacoustic experiments: the number of sample realizations must be selected by which the statistical estimate is determined in order that the measurement error be minimum. To do this, it is first necessary to determine the dependence of the measurement error on the number of sample realizations which are used in averaging.

It is natural to find the mean square error of statistical measurements ρ_θ (N) in the form:

$$\rho_\theta(N)=\langle[\hat{\theta}(\ell)-\theta(\ell/m_p)]^2\rangle. \tag{26}$$

Using (24)-(26) and assuming that any pair of measuring experiments leads to statistically independent sample realizations, we find:

$$\rho_\theta(N)=\frac{1}{N^2}\sum_{\kappa=1}^{N}d_\theta(\ell/m_\kappa,m_p)+ \\ +\left[\frac{1}{N}\sum_{\kappa=1}^{N}\theta(\ell/m_\kappa,m_p)-\theta(\ell/m_p)\right]^2. \tag{27}$$

We note that the first term in relation (27) is the standard deviation of the fluctuation error

$$d_\theta(N)=\frac{1}{N^2}\sum_{\kappa=1}^{N}d_\theta(\ell/m_\kappa,m_p),$$

and that the second term (in parentheses) is the bias of the estimate

$$S_\theta(N)=\frac{1}{N}\sum_{\kappa=1}^{N}\theta(\ell/m_\kappa,m_p)-\theta(\ell/m_p).$$

For further calculations let us consider the special case when

$$\left.\begin{aligned}d_\theta(\ell/m_\kappa,m_p)&=d_\theta(\ell,\varphi_\theta(\kappa,p)) \\ \theta(\ell/m_\kappa,m_p)&=\theta(\ell/m_p)\varphi_\theta(\kappa,p)\end{aligned}\right\} \tag{28}$$

which corresponds to the "multiplicative" dynamics of measuring experiments. Then for the relative total error (within the dimension of the probability characteristic)

$$\delta_\theta(N)=\frac{\rho_\theta^{1/2}(N)}{\theta(\ell/m_p)}, \tag{29}$$

with regard to the notation

$$\Delta_\theta=\frac{d_\theta^{1/2}(N)}{\theta(\ell/m_p)}, \tag{30}$$

we find:

$$\delta_\theta(N)=\left\{\frac{1}{N^2}\Delta_\vartheta^2\sum_{k=1}^{N}\varphi_\vartheta(k,p)+\left[\frac{1}{N}\sum_{k=1}^{N}\varphi_\theta(k,p)-1\right]^2\right\}^{1/2}. \tag{31}$$

Let us further assume that the dynamic experiment is characterized by no more than the quadratic dependence of functions $\varphi_\vartheta(k, p)$ and $\varphi_\theta(k, p)$ on the number of the sample realization k_u and on the number p of the true model m_p.

In other words, we assume that

$$\left.\begin{aligned}\varphi_\vartheta(k,p)&=a_\vartheta(k-p)+b_\vartheta(k-p)^2+1\\ \varphi_\theta(k,p)&=a_\theta(k-p)+b_\theta(k-p)^2+1\end{aligned}\right\} \tag{32}$$

Substituting (32) into (31), after simplification of the expression, we find:

$$\begin{aligned}\delta_\theta(N)=\Bigg\{\frac{1}{N}\Delta_\vartheta^2\Big[a_\vartheta\Big(\frac{N+1}{2}-p\Big)+b_\vartheta\Big[\frac{(N+1)(2N+1)}{6}-\\ -p(N-1)+p^2\Big]+\Big(a_\theta\Big(\frac{N+1}{2}-p\Big)+\\ +b\Big[\frac{(N+1)(2N+1)}{6}-p(N+1)+p^2\Big]\Big)^2\Bigg\}^{1/2}\end{aligned} \tag{33}$$

Examples of the functions $\delta_\theta(N)$, corresponding to different parameters contained in formula (33) in the case when the true model m_p corresponds to p = 0, are presented in Figure 5.

The following conclusions can be made from the results obtained:

-- the stronger the dynamic conditions of statistical measurements (increase of parameters a_v, b_v, a_θ and b_θ) are manifested, the less the value the optimum number of sample realizations assumes;

-- in some cases the optimum number of sample realizations is absent or rather it is equal to 1; this condition of the absence of a minimum measurement error corresponds to the case when any of the parameters a_ϑ, b_ϑ, a_θ, b_θ exceeds values 1-3;

-- the larger the number of the true model p, the larger the value the optimum number of sample realizations assumes and in this case the error value itself increases.

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GEOPHYSICS, ASTRONOMY AND SPACE

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ERRORS IN MEASURING THE ENVELOPE OF THE SPATIAL CORRELATION FUNCTION OF A QUASI-HARMONIC RANDOM FIELD

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 155-156

[Article by V. B. Galanenko and S. N. Maystrenko]

[Text] There are frequently cases in the practice of measuring the spatial correlation of random fields when the correlation function oscillates rapidly, which forces one to select a large number of reference points and in the final analysis complicates the entire experiment. Under these conditions it is feasible to turn to measurement of the envelope $A(\vec{x}, \vec{x}')$ and the phase $\phi(\vec{x}, \vec{x})$ of the complex correlation function (KKF) $\tilde{K}(\vec{x}, \vec{x}')$ whose real part coincides with the spatial correlation function. The functions $A(\vec{x}, \vec{x}')$ and $\varphi(\vec{x}, \vec{x}')$ vary comparatively slowly, which simplifies the measurement procedure. The errors of measuring $A(\vec{x}, \vec{x}')$ and $\varphi(\vec{x}, \vec{x}')$ are considered in this article.

Let us consider the space-time correlation function of a narrowband random acoustic field $K_{\Delta\omega}(\tau, \vec{x}, \vec{x}')$. One can show that

$$K_{\Delta\omega}(\tau,\vec{x},\vec{x}') = Re[B(\tau)g(\vec{x},\vec{x}')]\cos\omega\tau - Jm[B(\tau)g(\vec{x},\vec{x}')]\sin\omega\tau, \tag{1}$$

where $g(\vec{x}, \vec{x}')$ is a complex spatial correlation function and $B(\tau)$ is the envelope of the time correlation function. By substituting $\tau = 0$ and $\tau = \pi/2\omega$ into (1) and taking into account that $B(\pi/2\omega_0) \simeq B(0) = \sigma^2$, we find that the real part of the correlation function is $g_R(\vec{x}, \vec{x}') = K_{\Delta\omega}(0, \vec{x}, \vec{x}')$ and that the imaginary part is $g_y(\vec{x}, \vec{x}') = K_{\Delta\omega}(\pi/2\omega_0, \vec{x}, \vec{x}')$. Thus, by measuring the correlation function at two values of delay ($\tau = 0$ and $\tau = \pi/2\omega_0$), we can find the envelope $A = \sqrt{g_R^2 + g_y^2}$ and the phase $\Phi = \text{arc tg } g_y/g_R$.

Let us calculate the errors of measuring A and Φ, assuming that the estimates $\check{g}_R = \check{g}_y$ are unbiased and are determined by normal law with standard deviation σ^2 and cross-correlation coefficient r. After calculating the corresponding integrals, we arrive at the following relations. The bias of the estimates $\check{A}$ and $\check{\Phi}$ is equal to:

$$\Delta A = \frac{\sigma^2}{2A}\left(1 - \frac{r}{2}\sin 2\Phi\right) + \frac{3}{32}\frac{\sigma^4}{A^3}\left[1 + 2r\sin 2\Phi + \frac{r^2}{4}(3 + 5\cos 4\Phi)\right] \text{ при } \frac{\sigma}{A} \leq 1, \qquad (2)$$

$$\Delta\Phi = -\frac{\sigma^2}{A^2} r\cos 2\Phi - \frac{15}{4}\frac{\sigma^4}{A^4}\left[(1-r)^2\sin 4\Phi - (1-r)\frac{r}{2}\cos 4\Phi\right] \text{ при } \frac{\sigma}{A} \leq 0.3. \qquad (3)$$

The total errors of measuring A and Φ are equal to:

$$\Delta^2_{\Sigma A} = 2\sigma^2 - 2A\cdot\Delta A, \quad \Delta^2_{\Sigma\Phi} = g^2 - 2\Phi\cdot\Delta\Phi$$

where

$$g^2 = \left(\frac{\sigma}{A}\right)^2\left[1 - r(\sin 2\Phi + 2\Phi\cos 2\Phi)\right]$$

The graphs of the functions of bias errors and total errors are presented in Figures 1, 2, 3 and 4. The results of calculations by formulas (2) and (3) are plotted by the solid lines and the results of numerical integration are plotted by the dashed lines.

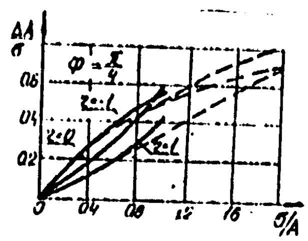

Figure 1

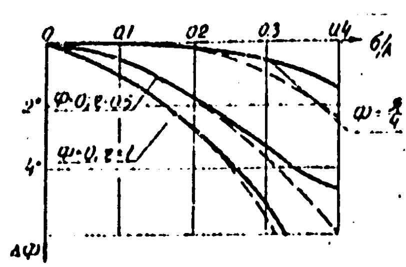

Figure 2

Analyzing the formula for $\Delta^2_{\Sigma A}$ jointly with (2), we find that $\Delta_{\Sigma A} \to g'\sqrt{1 - r/2}$ at $\sigma/A \to 0$ and that $\Delta_{\Sigma A} \to \sqrt{2}\sigma$ at $\sigma/A \to \infty$ $(A \to 0)$. This

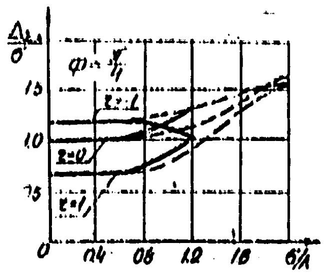

Figure 3

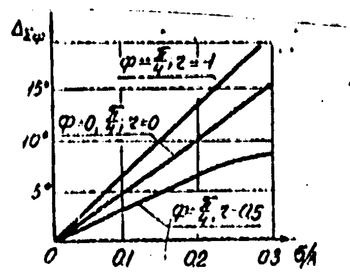

Figure 4

result and the graphs of Figure 3 show that the error in measuring the values of the envelope far exceed that of direct measurement of the values of the correlation function itself. The error in measuring phase Φ is considerably dependent on the values of phase and the correlation coefficient r and does not exceed 20° at $\sigma/A \leqslant 0.3$. The feasibility of experimental determination of the spatial correlation function by the envelope and phase of the complex correlation function confirms the estimates of the error values obtained.

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PROCESSING HYDROACOUSTIC SIGNALS BY USING THE ANALYTICAL PROPERTIES OF ESTIMATES OF THEIR INSTANTANEOUS FREQUENCY

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY KHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 157-164

[Article by S. A. Bachilo, V. B. Vasil'yev and G. M. Makhonin]

[Text] One of the main factors which make objective interpretation of the results of geophysical investigations considerably difficult is the transient nature of the level of hydroacoustic signals and also noise reflected from the investigated objects.

This factor, specifically, forces one in a number of cases to reject the use of optimum linear filtration in the receiving channels of hydroacoustic apparatus and to give preference to such methods and processing devices in which local restriction of the instantaneous values of signals is used. The latter devices include, specifically, ShOU systems and different variants of circuits for signal detection by the estimates of their instantaneous frequency.

The overall disadvantage of these devices is that their detection characteristics are considerably worse not only than those of devices with optimum linear filters, but also than those of typical detection channels.

Some possibilities of improving signal detection characteristics by estimates of the instantaneous frequency due to the use of analytical (according to terminology of [1]) properties of these estimates, specifically, the properties of their time derivatives and also one of the variants of devices for signal detection in which these possibilities are realized, are considered below.

Let us assume in the consideration that:

-- the random process

$$U(t) = U_m(t)\cos[\omega t + \varphi(t)] \quad (1)$$

subject to processing is narrowband and is the sum of the harmonic signal

$$S(t) = U_m(t)\cos[\omega_0 t + \varphi(t)] \qquad (2)$$

in Gaussian noise $\xi(t)$ with spectral density symmetrical with respect to frequency ω_0;

-- the estimate $\Delta\omega^*(t)$ of instantaneous frequency fluctuations is described by the functional:

$$\Delta\omega^*(t) = \int_{-\infty}^{\infty} h(t-\tau)\Delta\omega(\tau)\,d\tau \qquad (3)$$

This functional corresponds to a linear frequency detector with transfer function symmetrical with respect to frequency ω_0 and with constant parameters; $h(t)$ is the pulsed transfer characteristic of the output circuits of this detector. These circuits are essentially a low-frequency filter.

The process $\Delta\omega^*(t)$ at the output of this filter may be regarded as normal with a high degree of accuracy, and the same is also valid for the time derivatives of this process:

$$\Delta\omega^{*(i)}(t) = \int_{-\infty}^{\infty} \frac{\partial^i h(t-\tau)}{\partial t^i}\Delta\omega(\tau)\,d\tau, \quad i = 1, 2, 3\ldots \qquad (4)$$

The mean values and standard deviations of processes $\Delta\omega^*(t)$ and $\Delta\omega^{*(i)}(t)$ depend on the ratio of the mean square values of the signal $\sigma_c(t)$ and noise $\sigma_n(t)$, i.e.,

$$q(t) = \frac{\sigma_c(t)}{\sigma_n(t)} \qquad (5)$$

at the output of the frequency detector in the following manner: the mean values vary from zero in the absence of a signal ($q = 0$) and approach values of $\Delta\omega_c(t) = d\varphi_c(t)/dt$ and $\Delta\omega_c^{(i)}(t) = d^{i+1}\varphi_c(t)/dt^{i+1}$, respectively, with an increase of q (if $h(t)$ are selected so that $\Delta\omega^*(t)$ and $\Delta\omega^{*(i)}(t)$ are unbiased estimates); the standard deviations have the largest values in the absence of a signal and decrease sharply with an increase of q. Therefore, the values of the probabilities $P_0(q, t)$ and $P_i(q, t)$ of the presence of processes $\Delta\omega^*(t)$ and $\Delta\omega^{*(i)}(t)$ near $\Delta\omega_c(t)$ and $\Delta\omega_c^{(i)}(t)$, respectively, may serve as the criterion for making a decision about the presence or absence of a signal.

In this case, since the moment of the appearance of a signal is unknown beforehand, one must use the derivative $\Delta\omega^{*(i)}(t)$ of the order i in which the value of $\Delta\omega_c^{(i)}(t)$ is constant or hardly varies compared to the mean square value of the estimate $\Delta\omega^{*(i)}(t)$ at $q = 0$.

The use of estimates $\Delta\omega^{*(i)}(t)$ permits one to be freed of one of the main disadvantages of other methods of signal detection by the estimates of the instantaneous frequency -- significant deterioration of the detection characteristics with a decrease of signal frequency difference ω_c with respect to frequency ω_0.

Additional possibilities of improving the detection characteristics include the use of the properties of the combination of derivatives of different order due to estimates of the instantaneous frequency. This follows directly from the fact that in the general case the cross-correlation coefficients of estimate (3) and of its time derivatives (4) and also of derivatives of type (4) of different orders between each other are not identically equal to unity and, therefore, additional information about $\Delta\omega(t)$ is contained in $\Delta\omega^{*(i)}(t)$.

The decision about the presence of a signal when using the combination of derivatives of m- and n-order in the simplest cast can be made by the excess of the sum of probabilities $P_m(q, t), \ldots, P_n(q, t)$, i.e.,

$$L(q,t)=\sum_{i=m}^{n} P_i(q,t) \tag{6}$$

of the given threshold level.

This algorithm for detection of hydroacoustic signals is realized in a device whose block diagram is given in Figure 1, where the following is denoted: 1 -- band filter; 2 -- limiter; 3 -- frequency detector; 4_1-4_n -- differentiating links from first to n-th order, respectively; 5_1-5_n -- two-threshold amplitude discriminators; 6 -- adder; 7 -- low-frequency filter; 8 -- threshold stage. The protective circuit developed in the absence of input voltage is not shown in Figure 1.

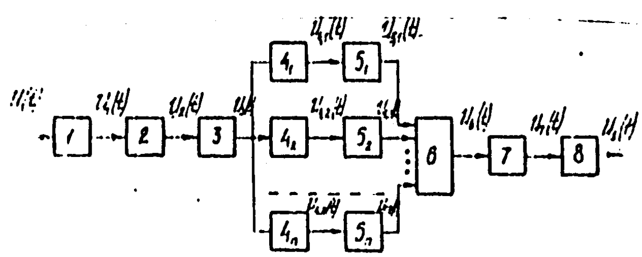

Figure 1. Signal Detection Device

The time diagrams which illustrate the operation of the main blocks of the device in Figure 1 are presented in Figure 2 as an example of signal detection with constant carrier frequency.

Realization of the input voltage U(t) is shown in Figure 2, a, realization of the output voltage of the frequency detector 3 is shown in Figure 2, b,

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realization of the output voltages of the differentiating circuits 4_1 and 4_2 of first- and second-order, respectively (the type of output voltages of differentiating circuits 4_3-4_n is similar to Figure 2, c and Figure 2, e) is shown in Figure 2, e, realization of the output voltages of amplitude discriminators 5_1 and 5_2 (the type of output voltages of discriminators 5_3-5_n is similar to that in Figure 2, d and Figure 2, f) is shown in Figure 2, f, realization of the output voltage of adder 6 is shown in Figure 2, g, realization of the output voltage of the low-frequency filter 7 is shown in Figure 2, h (level V_0 of response of the threshold stage 8 is shown by the dashed line in this same figure) and realization of the output voltage of threshold stage 8 is shown in Figure 2, i.

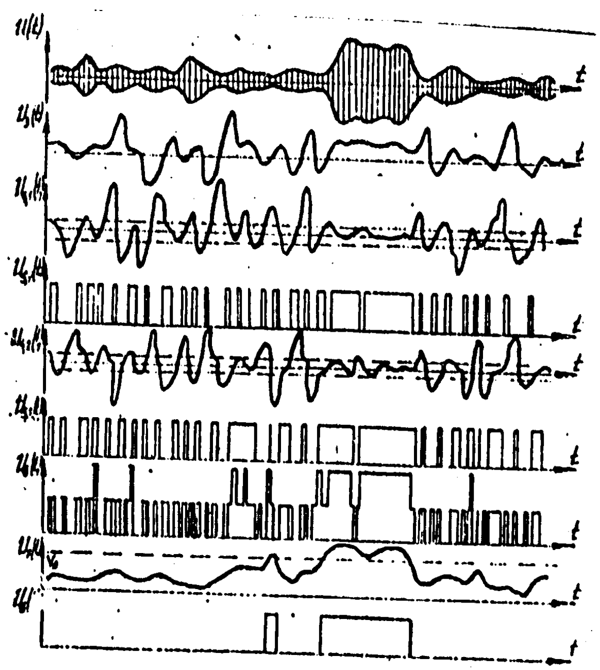

Figure 2. Time Diagrams

When determining the detection characteristics of the device in Figure 1, for simplicity of calculations it was assumed that:

1. A signal S(t) of type (2) at the input of the device has a constant amplitude U_{mc}, frequency ω_c and length T_c.

2. The band filter 1 has Gaussian amplitude-frequency characteristic $K_1(f)$, i.e.,

$$K_1(f) = e^{-\pi\left(\frac{f-f_0}{\Delta f_e}\right)^2}, \tag{7}$$

where f_0 is the central frequency of the filter and Δf_e is its energy bandpass, where $\Delta f_e \ll f_0$.

3. The spectral density of noise at the filter output 1 coincides with the square of its amplitude-frequency characteristic.

4. The output inertial circuits of the frequency detector 3 form a low-frequency filter with Gaussian amplitude-frequency characteristic:

$$K_3(f) = e^{-\pi\left(\frac{f}{\Delta f_3}\right)^2}, \tag{8}$$

where Δf_3 is the energy bandpass of this filter.

5. Amplitude discriminators 5_1-5_n have the transfer functions:

$$U_{5,i}(t) = \begin{cases} 1 \text{ at } |U_{4,i}(t)| \leq \Delta U_i \\ 0 \text{ at other } U_{4,i}(t) \end{cases} \tag{9}$$

6. The low-frequency filter 7 has a transfer coefficient equal to unity at $f = 0$ and an energy bandpass Δf_7, where $\Delta f_7 \ll \Delta f_3$.

7. The threshold stage 8 has the transfer function that

$$U_8(t) = \begin{cases} 1 \text{ at } U_7(t) \geq V_0 \\ 0 \text{ at } U_7(t) < V_0 \end{cases} \tag{10}$$

With the adopted assumptions, the probability distribution of instantaneous values of voltage $U_7(t)$ of the device in Figure 1 may be assumed as normal since the probabilities of correct detection D and of a false alarm P_{lt} are determined rather simply:

$$\begin{cases} P_{lt} = 1 - F\left(\frac{V_0 - m\{U_{7n}\}}{\sigma_{7n}}\right) \\ D = 1 - F\left(\frac{V_0 - m\{U_{7cn}\}}{\sigma_{7cn}}\right) \end{cases}, \tag{11}$$

where $F(x) = \frac{1}{\sqrt{2\pi}}\int_{-\infty}^{x} e^{-\frac{t^2}{2}}dt$ is the probability integral, $m\{U_{7n}\}$ and $m\{U_{7cn}\}$ are mean values and σ^2_{7n} and σ^2_{7cn} are the standard deviations of low-frequency voltage fluctuations $U_7(t)$ at the low-frequency filter output 7 when acting on the input of the device in Figure 1 by noise alone or by a mixture of a signal with noise, respectively.

It is obvious that

$$m\{U_4(t)\} = \sum_{i=1}^{n} m\{U_{4,i}\}, \tag{12}$$

and the standard deviations σ_4^2 is approximately determined by the expression

$$\sigma_4^2 \approx 4\Delta f_\phi \int_0^\infty B_6(\tau)\,d\tau, \tag{13}$$

where

$$m\{U_{4,i}\} = 2F\left(\frac{\Delta_i}{\kappa_i q}\right) - 1, \tag{14}$$

$B_0(\tau)$ is the correlation function of voltage $U_5(t)$, $\frac{\Delta_i}{\sigma_{3,i}}$, are the normalized threshold voltages of the amplitude discriminators, are coefficients dependent on the number of the derivative i and on the signal/noise ratio q at the filter output 1, $\sigma_{3,i}$ are the standard deviations of voltages $U_{\mu,i}(t)$ and $B_7(\tau)$ is the correlation function of voltage $U_3(t)$. Correlation function $B_6(t)$ is determined by the expression:

$$B_6(\tau) = \sum_{i=1}^{n}\sum_{j=1}^{n}\left[\int_{-\Delta_i}^{\Delta_i}\int_{-\Delta_j}^{\Delta_j} W_2(U_{4,i}; U_{4,j})\,dU_{4,i}\,dU_{4,j} - m\{U_{4,i}\}\,m\{U_{4,j}\}\right], \tag{15}$$

where $W_2(U_{4,i}; U_{4,j})$ is the two-dimensional probability distribution density of expressions $U_{4,i}(t)$ and $U_{4,j}(t+\tau)$ or with representation of $W_2(U_{4,i}; U_{4,j})$ by a Kramer series

$$B_6(\tau) = 4\sum_{i=1}^{n}\sum_{j=1}^{n}\sum_{\ell=1}^{\infty} F^{(2\ell)}\left(\frac{\Delta_i}{\kappa_i q}\right) F^{(2\ell)}\left(\frac{\Delta_j}{\kappa_j q}\right) \frac{R_{4,ij}^{2\ell}(\tau)}{(2\ell)!}, \tag{16}$$

where

$$R_{4,ij}(\tau) = \begin{cases} \dfrac{(-1)^i B_3^{(i+j)}(\tau)}{\sqrt{(-1)^{i+j} B_3^{(2i)}(0)\, B_3^{(2j)}(0)}} & \text{at } i \neq j \\[2ex] \dfrac{B_3^{(2i)}(\tau)}{B_3^{(2i)}(0)} & \text{at } i = j \end{cases}$$

Specifically, when approximating the energy spectrum G(f) of the instantaneous frequency, corresponding to [2], i.e.,

$$G(f) = [illegible] \tag{17}$$

the correlation function $B_3(\tau)$ has the form

$$B_3(\tau) = [illegible] \tag{18}$$

and coefficients $K_i(q)$ are determined by the expression

$$K_1(q) = [illegible] \tag{19}$$

where $\alpha = \Delta f_3/\Delta f_e$.

Figure 3 illustrates the results of calculating the detection characteristics of the device in Figure 1 and the typical channel and signal detection device by estimates of the instantaneous frequency found at $P_{nt} = 10^{-2}$ and at different relative frequency differences $\nu = \Delta f_c/\Delta f_e$ of the signal frequency. The detection characteristics of the device in Figure 1 and of the typical channel are essentially independent of the value of ν and are represented by graphs 2 and 1, respectively. The detection characteristics of the device, based on the use of estimates $\Delta\omega^*(t)$, are represented by graphs 3, 4 and 5 and by the corresponding frequency differences $\nu = 0.38$, 0.36 and 0.34. It was assumed during calculations for the device in Figure 1 that n = 2, $r_1 = 0.5$ and $r_2 = 0.6$. The first 85 terms of the series were taken into account in (16) when determining $B_6(\tau)$.

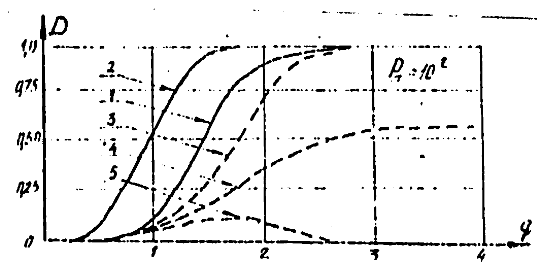

Figure 3. Detection Characteristics

As can be seen from the graphs of Figure 3, the use of the properties of the combination of the first two derivatives of estimates of the instantaneous frequency permits one to improve the detection characteristics of the signals, especially with small relative frequency difference.

It is also obvious from Figure 3 that the device in Figure 1 has worse detection characteristics than the typical channel, but, unlike the latter, it provides constant probability of false alarms during transient input process U(t) so that this deterioration of these characteristics is a plateau for stabilization of the probability of false alarms.

Improvement of the detection characteristics of the device in Figure 1 compared to the device for signal detection by estimates $\Delta\omega^*(t)$ is clearly illustrated by Figure 4, in which photographs of the output voltage oscillograms of these devices are presented (the upper oscillograms on the photographs correspond to the device in Figure 1), found with signal/noise ratios at the inputs of the device of 1.5 and 2.0 (photographs a and b, respectively) and with relative signal frequency differences $\nu = 0.2$.

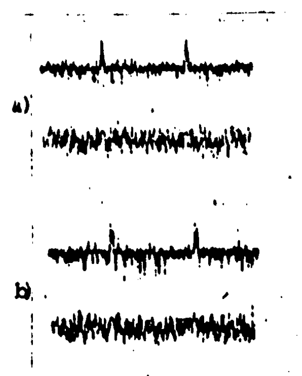

Figure 4. Voltage Oscillograms

Hydroacoustic signal processing devices (similar in structure to the device in Figure 1), based on the use of the analytical properties of estimates $\Delta\omega^*(t)$ makes it technically simple to accomplish separation of signals which differ by the type of frequency modulation, for example, to provide joint operation of several sonar devices in a single water basin. Signal separation is more effective, the more significantly derivatives $\Delta\omega_c^{(i)}(t)$ differ in value.

Photographs of the output voltage oscillograms of the signal separation device (lower oscillograms) and also oscillograms of the output voltages of the frequency detector of this device with separation of signals with increasing and decreasing linear frequency modulation having derivatives $\Delta\omega_c'$ opposite in sign on a noise background. Photographs a and b were obtained with signal/noise ratios of 1.5 and 2.0, respectively. As can be seen from Figure 5, separation of these signals is sufficiently effective even at signal/noise ratios on the order of 1.5-2.0.

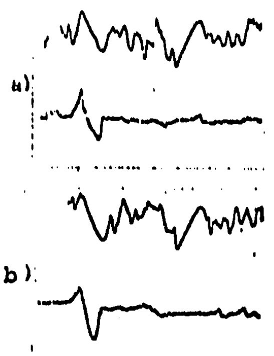

Figure 5. Voltage Oscillograms

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GEOPHYSICS, ASTRONOMY AND SPACE

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EFFECT OF REFERENCE SIGNAL TRUNCATION ON THE NOISE STABILITY OF AN OPTIMUM DETECTOR

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY KHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 169-170

[Article by V. I. Chaykovskiy]

[Text] Let us estimate the degree of variation of signal-noise ratio due to use of a reference signal $r_T(t)$ $(t \in T)$, acting in a matched finite interval with detection of a precisely known energy E_S signal $S(t)$ on a normal tinted noise background $x(t)$ with correlation function $K(t, \tau)$.

With unlimited processing interval, the signal-noise ratio has the form

$$q_{\infty} = \frac{1}{E \tau} \int_{-\infty}^{\infty} \frac{|S(\omega)|^2}{P(\omega)} d\omega, \tag{1}$$

where $S(\omega)$ and $P(\omega)$ are the signal and noise spectra, respectively.

With a limited processing interval, this same ratio will be equal to

$$q_T = \frac{1}{E \tau} \frac{\left| \int_{-\infty}^{\infty} S(\omega) R_T^*(\omega) d\omega \right|^2}{\int_{-\infty}^{\infty} P(\omega) |R_T(\omega)|^2 d\omega}, \tag{2}$$

where $R_T(\omega)$ is the truncated reference signal spectrum.

Variation of the signal-noise ratio with truncation of the reference signal may be estimated by the value of the loss coefficient of noise stability

$$K = \frac{q_{\infty} - q_T}{q_T}.$$

If the interval of reference signal truncation T is limited below by the interval of existence of a useful signal $T_S \subset T$, the loss coefficient with regard to (1) and (2) may be represented by the expression

$$K = \frac{\int_{-\infty}^{\infty} P(\omega)|R_T(\omega)|^2 d\omega}{\int_{-\infty}^{\infty} P(\omega)|R(\omega)|^2 d\omega} - 1. \quad (3)$$

Reference signal truncation leads to smoothing of its spectrum $R(\omega)$. Therefore, the truncated signal spectrum $R_T(\omega)$ may be represented in the form of the sum

$$R_T(\omega) = R(\omega) + G(\omega),$$

where $G(\omega)$ is some complex function of deviation in the general case. Consequently,

$$|R_T(\omega)|^2 = |R(\omega)|^2 + R(\omega)G^*(\omega) + R^*(\omega)G(\omega) + |G(\omega)|^2$$

or, since the square of the modulus of the spectrum of the deviation function is small compared to the product $|R(\omega)||G(\omega)|$

$$|R_T(\omega)|^2 \approx |R(\omega)|^2 + R(\omega)G^*(\omega) + R^*(\omega)G(\omega).$$

Substitution of (4) into (3) with regard to the fact that the reference signal spectrum is clearly determined through the signal and noise spectra by the expression

$$R(\omega) = E_S \frac{S(\omega)}{P(\omega)}.$$

Having used the Bunyakovskiy-Shvarts inequality and having amplified it by replacing the energy of the real and imaginary components with total energy, with regard to (1), we find

$$K \leq \frac{4}{q_\infty E_S}\left(\frac{1}{2\pi}\int_{-\infty}^{\infty}|S(\omega)|^2 d\omega\right)^{1/2}\left(\frac{1}{2\pi}\int_{-\infty}^{\infty}|G(\omega)|^2 d\omega\right)^{1/2}.$$

The first expression in parentheses is the total energy of the useful signal E_S and the second expression by definition coincides with the energy of the deviation function or, which is the same thing, with the energy of continuation of the extreme reference signal outside the processing interval ΔE_T. Thus,

$$\kappa \leqslant \frac{1}{q_{1m}} \left(\frac{\Delta E_r}{E_s} \right)^{1/2}.$$

Consequently, the degree of violation of optimum detection with amplification of the reference signal is determined by the ratio of the energy of reference signal truncation ΔE_r to the total energy of the useful signal E_s and is inversely proportional to the extreme value of the signal-noise ratio $q_{\Delta 0}$. This circumstance qualitatively justifies the proximity of quasi-optimum detector characteristics to a truncated reference signal and of an optimum detector without restrictions in a number of practically important cases.

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GEOPHYSICS, ASTRONOMY AND SPACE

UDC 621.391.2

THE INFLUENCE OF THE DOPPLER EFFECT ON THE RESPONSE OF A DISCRETE MATCHED FILTER

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 p 175

[Article by K. P. L'vov]

[Text] The response of a discrete matched filter as a discrete finite convolution of lattice functions is considered. The results of modeling discrete matched filtration (cross-correlation processing), which was carried out on the basis of the algorithm of fast Fourier transform and using the discrete finite analytical concept of the pulsed function (reference signal) [1] are presented in the figure. The mathematical model of the emitted signal was a signal with square-wave envelope and linear frequency modulation

$$S(t) = \cos[2\pi(f_0 t + \frac{F}{2T_c} t^2)]$$

with the following values of parameters $T_c = 1.0$ sec, $F = 150$ Hz and $f_0 = 2,700$ Hz. The Doppler effect was taken into account as a frequency shift. The dependence of response amplitude $S_{vykh}[kT]$, time shift $|kT|/T_c$ and of constant time resolution FT_r on the value of $2|v|/c$, where c is the speed of propagation of fluctuations in the medium and v is the radial component of the rate of displacement, are presented in the figure. The theoretical functions were taken from [2] and are plotted in Figure 1 by the solid lines.

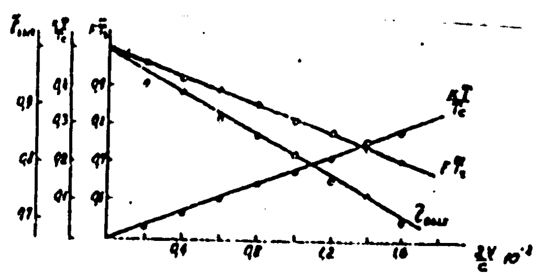

Figure 1. Dependence of Response Characteristics on Value

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GROPHYSICS, ASTRONOMY AND SPACE

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AN ALGORITHM FOR PROCESSING BINARY-QUANTIFIED SIGNALS

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 177-178

[Article by B. I. Pakhomkin and K. V. Filatov]

[Text] When detecting signals with unknown initial phase, devices which separate the quadrature components of the signal and their functional transformation -- squaring and addition -- are usually employed in hydroacoustics. However, technical realization of these devices both in analog and in digital form encounters a number of difficulties. These difficulties are related to the following requirements: high identity of fulfilling the quadrature channels and large dynamic range of signal and squaring multiplication devices and performing a considerable number of necessary computing operations (multiplications and additions). Therefore, it is feasible to change the algorithm for quadrature processing to simplify performance of multiplication, addition and nonlinear transformation operations.

Performing arithmetic operations is simplified to the greatest degree when using binary quantification of the amplitude of both input and reference oscillation, but in this case the required type of nonlinear functional transformation of output signals of the quadrature channels in which invariance of the processing algorithm to the random initial phase of the signal is achieved is unknown.

The output signals of the quadrature channels $Z_c(\tau_u)$ and $Z_s(\tau)$ are written in the form

$$Z_{c,s}(\tau) = \int_0^T sign\{x(t)\} \cdot sign\begin{Bmatrix} S_o(t-\tau) \\ S_o^*(t-\tau) \end{Bmatrix} dt, \qquad (1)$$

where $S_o(t)$ and $S_o^*(t)$ are the reference oscillations conjugate by Hilbert. Let us find the function $Z(\tau)$ at the moment of ending of the input signal from the initial phase of the processed signal φ. Substituting the binary-quantified signal with phase modulation $\Phi(t)$, represented in the form of a series [1], into (1) and performing the required mathematical operations, we find

$$Z(\varphi, \tau) = \frac{8T}{\pi^2} \sum_{k=1}^{\infty} \frac{\cos[(2k-1)\varphi]}{(2k-1)^2}. \tag{2}$$

Expression (2) does not depend on the law of signal phase modulation and is the expansion of a symmetrical sawtooth function into a Fourier series. Therefore, the following functional transformations of the values $Z(\varphi, \tau)$ and $Z(\varphi - \pi/2, \tau)$ -- calculation of the functions of the modulus with subsequent addition -- must be carried out to achieve invariance of the processing algorithm to the random initial phase. The modified algorithm for processing binary-quantified signals may be written in the form

$$Y(\tau) = |Z_s(\tau)| + |Z_c(\tau)|. \tag{3}$$

An experimental check of algorithm (3) was made. A model of a discrete matched filter for LChM [Linear frequency modulation] of signals with base value B = 48 was developed. A single-channel shift register was used as the delay device due to the use of binary quantification of the input oscillation amplitude. Since the weight coefficients, according to (1), assume only two values +1 and -1, the multiplication operation was carried out by the simplest logic circuits.

The measured value of relative fluctuations of the filter response amplitude with random variations of the initial signal phase comprised 14 percent. The noise stability of the filter was analyzed by the detection curve, measured according to the method outlined in [2]. The value of the threshold signal comprises 8.7 with the probability of a false alarm equal to 10^{-3} and the probability of correct detection of 0.9. It follows from comparison of the threshold signals of the investigated device and the filtration circuit to the quadrature channels and limitation of the process at the input that the threshold signal increases insignificantly -- by 2.9 dB.

The processing algorithm (3) may be used for signals with different laws of frequency and phase modulation. The advantages of technical realization of this algorithm are obvious. Remultiplication and functional transformation operations both in the digital and analog variants of the signal processing devices are considerably simplified.

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GEOPHYSICS, ASTRONOMY AND SPACE

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FAST METHOD OF REALIZING CORRELATION SIGNAL PROCESSING BY USING A DIGITAL COMPUTER

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 179-180

[Article by K. B. Krukovskiy Sinevich and V. V. Mikhaylovskiy]

[Text] It is known that digital correlation processing of signals is completing calculations of type:

$$y(\tau_i)=\frac{1}{N}\sum_{j=i}^{N+i}x(t_j)\cdot u(t_j-\tau_i),\quad i=1,2\ldots M-N, \tag{1}$$

where x(t), y(τ) and u(t - τ) are the input, output and reference signals, respectively, M is the number of readings of the input signal and N is the number of readings of the reference signal.

The developer is most frequently interested in processing devices which operate in real time, i.e., in a mode when the time required to calculate N points y(τ) would be equal to N quantification intervals of the input process x(t). The speed of the calculator does not meet these requirements for some problems; therefore, development of algorithms equivalent to (1), but which permit a reduction of the volume of calculations, is of sufficient interest. The most well known among rapid algorithms are BPF [expansion unknown] [1] and algorithms which use theoretical-numerical transformations.

Yet another method of realizing correlation convolution of type (1), which permits a reduction in the volume of calculations by reducing the number of multistage multiplication operations is proposed in this paper.

It is known that digital machines operate with numbers represented in binary code. Having represented each value of the reference signal in the form of a sum by powers of 2, for expression 1, we find:

$$y(\tau_i)=\frac{1}{N}\sum_{j=i}^{N+i}\sum_{l=0}^{R-1}x(t_j)\cdot 2^{l}\cdot a_{jil}, \tag{2}$$

where R is the digit capacity of the reference signal and coefficient a_{jiL} determines the value of the reference signal sample $U(t_j - \tau_i)$ written in the L-th digit.

One may note during analysis of expression (2) that only coefficients a_{jiL} will be different for all values of τ_i $(1 < i < M - N)$ and since these coefficients may assume only two values "0" or "1," one may use the same values of $x(t_j) \cdot 2^L$ to form the different values of $y(\tau_i)$. The proposed method can be illustrated by the approximate graph (Figure 1).

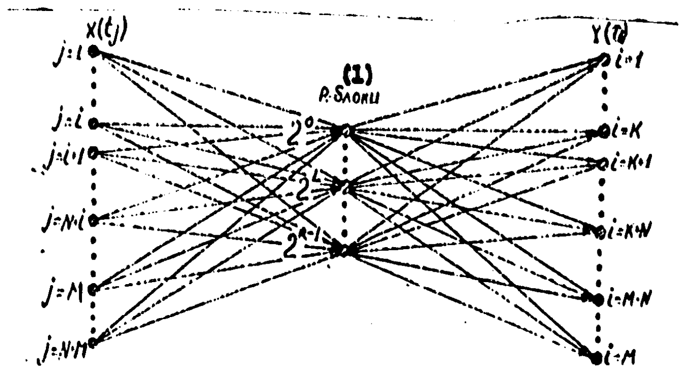

Figure 1. Signal Graph of Method

KEY: 1. R-blocks

The reading of the input signal (the left points of the graph) is multiplied by R weight coefficients, which are different powers of the number 2, and are entered into the intermediate memory (R-blocks). The value of each R-block is placed parallel to the N adders through "AND" elements, controlled by coefficients a_{jiL}. When using adders with storage, the latter may also perform the functions of internal storage for storing the values of $y(\tau)$. It should also be noted that there is no need in the memory for storage of N readings of the input process when using the given processing method, since these readings may be processed as they arrive. The internal storage of the processing device itself should then contain $N \cdot R$ bits or N R-digit words for storage of coefficients which characterize the reference signal sample, while the time of calculating N points of function $y(\tau)$ will be determined by the time required to perform $N \cdot (R - 1)$ shift operations and the same number of addition operations (when using N parallel adders), which is equivalent to a total of N multiplication operations.

Thus, the proposed processing method has an advantage in speed compared to algorithms based on BPF and $\log_2 N$-fold and approximately identical design complexity and may be recommended for processing sonar signals.

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GEOPHYSICS, ASTRONOMY AND SPACE

UDC 778.4:534.8

THE PROBLEM OF DIGITAL PROCESSING OF SPATIAL ACOUSTIC SIGNALS

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY KHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 181-182

[Article by V. V. Gritsyk, E. R. Zlatogurskiy, V. V. Koshevoy, V. N. Mikhaylovskiy and S. A. Soroka]

[Text] Methods of holographic recording of the reflected acoustic field and digital image restoration by using a specialized real-time digital computer are one of the most promising among known methods of constructing acoustic images.

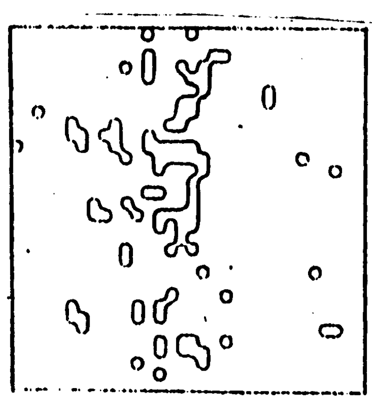

Figure 1. Restored Acoustic Image of Rectilinear Emitter

This paper is devoted to modeling the process of restoration and identification of the acoustic image for an experimentally recorded diffraction field of an object on a universal digital computer [1]. The steps of processing the image of a rectilinear emitter are presented in Figures 1, 2 and 3, a and b.

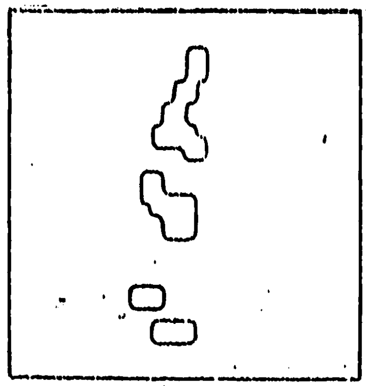

Figure 2. Acoustic Image of Rectilinear Emitter After Correction

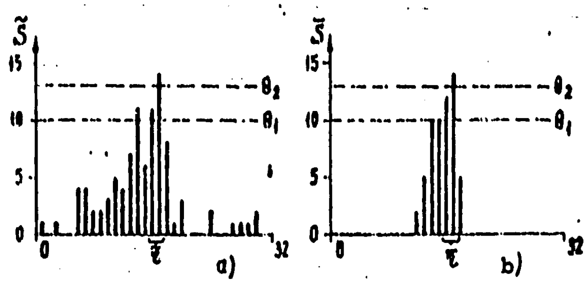

Figure 3. Spectrum of Image of Rectilinear Emitter: a -- after restoration; b -- after correction

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GEOPHYSICS, ASTRONOMY AND SPACE

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PROCESSING ONE CLASS OF HYDROACOUSTIC IMAGES IN REAL TIME

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 183-184

[Article by V. V. Gritsyk, A. Yu. Lutsyk, V. N. Mikhaylovskiy and G. T. Cherchyk]

[Text] Let us consider processing of one class of hydroacoustic images obtained on a receptor field of dimension n x n, selecting as the informative features elements of the internal structure of the images [1] and using the method of making the algorithm unparallel in processing the information.

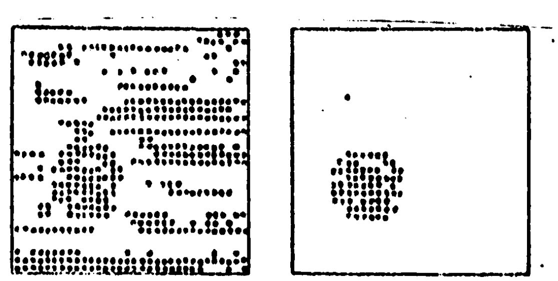

Figure 1. Image of Circle on Receptor Field of Dimension 32 x 32: a -- under noisy conditions; b -- after restoration

Let there be given an image-word of length n^2

$$v_i = (a_{i1}, a_{i2}\, a_{i3} \ldots a_{in^2}),$$

where the letters

$$a_{ik} \in E = \{0,1\}.$$

Let us determine the image spectrum as a word of length n

$$v_i^s = (a_{i1}^s \, a_{i2}^s \, a_{i3}^s \ldots a_{in}^s),$$

where the letters

$$a_{ii} \in E = \{0, 1, 2, \ldots, n\} \quad \text{and} \quad a_{ii}^s = \sum_{q=0}^{n-1} a_{ii+q}$$

The image v_i belongs to the class of images k if the letters a_{ir}^s of the spectrum are determined according to the expression

$$a_{ii,k} = 2\sqrt{R^2 - [R - (k - \tfrac{1}{2})]^2}.$$

An example of the image of a circle on a noisy background on a receptor field of dimension 32 x 32 is presented in Figure 1, a. This same image after restoration is presented in Figure 1, b. An image of an ideal circle at $n > \infty$ and the image of a circle with diameter D = 10 is presented in Figure 2, a. The restored image of a circle of diameter D = 10, distorted by noise, is presented by the dots in the same figure. The spectra v_i^s and $\tilde{v}_i^s$, respectively, are presented for these images in Figure 2, b. To identify the images belonging to class k, let us determine

$$|v_i^s - \tilde{v}_i^s| = |(a_{i1}^s \, a_{i2}^s \, a_{i3}^s \ldots a_{in}^s) - (\tilde{a}_{i1}^s \, \tilde{a}_{i2}^s \, \tilde{a}_{i3}^s \ldots \tilde{a}_{in}^s)| =$$
$$= (|a_{i1}^s - \tilde{a}_{i1}^s| \, |a_{i2}^s - \tilde{a}_{i2}^s| \, |a_{i3}^s - \tilde{a}_{i3}^s| \ldots |a_{in}^s - a_{in}^s|) = (\Delta_{i1} \, \Delta_{i2} \, \Delta_{i3} \ldots \Delta_{in})$$

Let Δ be some previously given number whose selection depends on the adopted model of the noise effect.

Then, if for all values of k,

$$\Delta_{ik} \leq \Delta,$$

the image $a_i \in k$.

It is obvious from Figure 2 that one may select $\Delta = 2, 1$, which characterizes the effect of twofold distortions on the image of the circle itself, for pattern recognition.

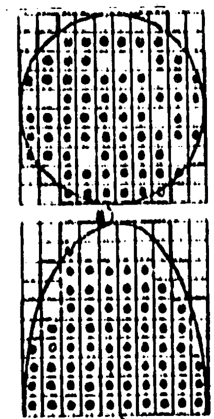

Figure 2. Spectra of Images: solid line -- of ideal image; dashed line -- of restored image

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GEOPHYSICS, ASTRONOMY AND SPACE

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THE SPECTRAL-CORRELATION STRUCTURE OF TRANSIENT MODELS OF RANDOM HYDROACOUSTIC PROCESSES

Novosibirsk TRUDY VOS'MOY VSESOYUZNOY SHKOLY-SEMINARA PO STATISTICHESKOY GIDROAKUSTIKE in Russian 1977 signed to press 14 Dec 77 pp 187-189

[Article by N. N. Vishnyakova, V. A. Geranin, A. N. Prodeus and G. D. Simonova]

[Text] The spectral-correlation characteristics of several models of continuous transient random processes (NSP) which satisfactorily approximate real random processes (SP) of hydroacoustic origin, are presented below. The basis of the calculation are the definitions used in [1] and [2].

The Response of a Parametric Determined Linear System to a Steady Random Effect

Let us be given a system with complex frequency characteristic $H_\varphi(\omega, t)$. The spectral-correlation characteristics of this model of NSP are as follows:

$$K(\tau,t) = \int_{-\infty}^{\infty} H_\varphi(\omega,t+\tau)\,\overline{H}_\varphi(\omega,t)\,G_\xi(\omega)\,e^{j\omega\tau}\,d\omega,$$

$$\Phi(\omega,t) = \frac{1}{2\pi}\int_{-\infty}^{\infty} \Gamma(\nu,\omega)\,\overline{H}_\varphi(\nu,t)\,G_\xi(\nu)\,e^{j(\omega-\nu)t}\,d\nu,$$

$$Q(\Omega,\omega) = \frac{1}{4\pi^2}\int_{-\infty}^{\infty} \Gamma(\nu,\omega)\,\overline{\Gamma}(\nu,\omega+\Omega)\,G_\xi(\nu)\,d\nu,$$

$$\chi(\tau,\Omega) = \frac{1}{2\pi}\iint_{-\infty}^{\infty} H_\varphi(\nu,t+\tau)\,\overline{H}_\varphi(\nu,t)\,G_\xi(\nu)\,e^{j(\Omega t+\nu\tau)}\,d\nu\,dt.$$

A periodically transient SP is:

$$X(t) = \varphi(t)\,\xi(t),$$

where

$$\varphi(t) = \sum_{n=-\infty}^{\infty} S_n\,e^{jn\Omega_0 t};$$

One can show that

$$
\begin{aligned}
{}_T\Phi(\omega, t) &= \frac{T}{2\pi}\,\mathrm{rect}\left(\frac{t}{T}\right)\int_{-\infty}^{\infty} S_a\left(\frac{\nu T}{2}\right) G(\omega-\nu)\, e^{j\nu t}\, d\nu, \\
{}_T Q(\Omega, \omega) &= \frac{T^2}{4\pi^2}\int_{-\infty}^{\infty} S_a\left(\frac{\nu+\Omega}{2}T\right) S_a\left(\frac{\nu T}{2}\right) G(\omega-\nu)\, d\nu, \\
{}_T \chi(\tau, \Omega) &= \frac{T^2}{4\pi^2} R(\tau)\int_{-\infty}^{\infty} S_a\left(\frac{\nu T}{2}\right) S_a\left(\frac{\Omega-\nu}{2}T\right) e^{j(\Omega-\nu)\tau}\, d\nu.
\end{aligned}
\tag{3}
$$

Of special applied interest is modification of model (1), corresponding to X(t) with

$$2\tau_{H.K.} < T. \tag{4}$$

In this case the following equalities are valid

$$
\begin{aligned}
{}_T K(\tau, t)\,\mathrm{rect}\left(\frac{t}{T - 2\tau_{H.K.}}\right) &= R(\tau), \\
{}_T \Phi(\omega, t)\,\mathrm{rect}\left(\frac{t}{T - 2\tau_{H.K.}}\right) &= G(\omega).
\end{aligned}
$$

In other words, with condition (4), the correlation function (2) and spectrum (3) of NSP (1) in the interval $-T/2 + \tau < t < T/2 - t$ does not differ from the correlation function and the spectrum of the steady SP X(t). N. A. Zheleznov called this NSP quasi-steady.

If

$$\tau_{H.K.} \ll T,$$

then

$$
\begin{aligned}
{}_T K(\tau, t) &\approx R(\tau), &(5) \\
{}_T \Phi(\omega, t) &\approx G(\omega), &(6)
\end{aligned}
$$

i.e., $K(\tau, t)$ and $\Phi(\omega, t)$ coincide with $R(\tau)$ and $G(\omega)$, respectively, on essentially the entire interval $(-T/2 + T/2)$.

The correlation function in frequency and the difference argument function in time and frequency of this quasi-steady process have the form:

$$
\begin{aligned}
{}_T Q(\Omega, \omega) &\approx \frac{T}{2\pi}\, S_a\left(\frac{\Omega T}{2}\right) G(\omega), \\
{}_T \chi(\tau, \Omega) &\approx \frac{T}{2\pi}\, S_a\left(\frac{\Omega T}{2}\right) R(\tau).
\end{aligned}
$$

the correlation function of the NSP x(t) has the form:

$$K(\tau,t) = \sum_{n=-\infty}^{\infty} {}_nR(\tau)\, e^{jn\Omega_0 t},$$

and

$${}_nR(\tau) = \sum_{m=-\infty}^{\infty} s_m \bar{s}_{m-n} R_\varepsilon(\tau)\, e^{-j(m-n)\Omega_0\tau}.$$

The spectrum

$$\varphi(\omega,t) = \sum_{n=-\infty}^{\infty} {}_nG(\omega)\, e^{jn\Omega_0 t},$$

where

$$G(\omega) = \sum_{m=-\infty}^{\infty} s_m \bar{S}_{m-n} G_\varepsilon[\omega + (m-n)\Omega_0]$$

One can show that

$$Q(\Omega,\omega) = \sum_{n=-\infty}^{\infty} G(\omega)\,\delta(\Omega + n\Omega_0)$$

and

$$\chi(\tau,\Omega) = \sum_{n=-\infty}^{\infty} {}_nR(\tau)\,\delta(\Omega + n\Omega_0).$$

Segment of Steady SP

The correlation function of the NSP of type:

$${}_TX(t) = \mathrm{rect}\left(\frac{t}{T}\right) X(t), \qquad (1)$$

where X(t) is the SSP [Steady random process] with correlation function $R(\tau)$ equal to

$${}_TK(\tau,t) = \mathrm{rect}\left(\frac{t}{T}\right)\mathrm{rect}\left(\frac{t+\tau}{T}\right) R(\tau) \qquad (2)$$

Let us emphasize that the spectral components whose frequency difference is a multiple of $2\pi/T$ are uncorrelated.

If X(t) is white noise, the approximate equalities (5) and (6) change to precise equalities.

Incidentally, if the inequality inverse to (4) is valid, only the following relations remain in force

$$\overline{\langle X_{T}(t)\rangle} = \overline{\langle X(t)\rangle}\ \mathrm{rect}\left(\frac{t}{T}\right),$$

$$D[X_{T}(t)] = D[X(t)]\ \mathrm{rect}\left(\frac{t}{T}\right).$$

The correlation function $K(\tau, t)$ and the spectrum $\phi(\omega, t)$ will depend on t over the entire interval (-T/2, +T/2).

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